UNIVERGE NEAX 2000 IPS

INTERNET PROTOCOL SERVER

Request For Proposal (RFP)
Reference Guide

May 2006

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The *UNIVERGE* NEAX 2000 IPS Request For Proposal (RFP) Reference Guide has been developed to provide technical information on the *UNIVERGE* NEAX 2000 IPS, which is required when responding to the majority of questions asked on Request For Proposals. The information provided has been compiled from a variety of available documentation and has been consolidated into a single manual. For this reason, it may be necessary to gather additional information from other supporting documentation to fulfill the Request For Proposal (RFP) requirements.

Information concerning questions not covered in this guide, corrections and/or comments are most welcome and should be sent to:

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Chapter 1 Introduction

Overview of NEC

NEC Corporation was founded in 1899. NEC worldwide is built on a strong tradition and global heritage. NEC was created out of a joint venture between Western Electric from America and a Japanese investment group. Japan's first joint venture combined the technology and manufacturing techniques of the Bell Company with the vision of Japanese investors to form one of the oldest, established telecommunications companies in the world. NEC is one of only a small number of companies within the world that have successfully pioneered technology and delivered products within every major evolution of business communications systems.

NEC, recognized as a worldwide leader in high technology, is one of the few companies capable of offering a full spectrum of products and systems in computers, communications, and semiconductor devices.

NEC first established a United States presence over 40 years ago when it opened a sales office in New York in 1963. Since that time, NEC has broadened its operations in the United States by expanding into manufacturing, research and software development operations, by employing 7,000 people and by establishing extensive marketing, sales and service networks nationwide with revenues exceeding $5.9 billion.

In 1993, NEC introduced a new corporate logo: NEC. The logo represented the beginning of a dynamic era in the life of a company with a distinguished past. The changing dynamics of domestic and international markets mandate that the world's leading companies evolve with those changes. In response to this evolution, NEC is building a more responsive business based on the steadfast foundation of the traditional NEC business philosophy. NEC's organization has become a recognized leader in linking people and information through technology. The logo evolved from the simple abbreviation of Nippon Electric Company, in the late 1890.s, into a dynamic symbol featuring the traditional NEC letters with bold, sharp lines expressing technology and confidence, combined with gentle curves that are associated with humanity and friendliness. The logo has a holistic design that each letter expresses an independent character and at the same time represents the harmony of the logo as a whole or unit. NEC blue has been designated as the corporate color. This subtle blue was selected to symbolize human intelligence, life, and the protection of the environment.

The Invention Age

NEC began its solid tradition with some of the first manual telecommunications systems in Japan. During the Invention Age, NEC had achieved the following:

- In 1900 - NEC began manufacturing its own products
- In 1903 - NEC manufactured the first battery phone in Japan
- In 1923 - NEC entered the radio transmission field
- In 1927 - NEC began automatic switching manufacturing
- In 1938 - NEC began manufacturing crossbar switching systems

The Industrial Age

The Industrial Age in America brought great strides in business communication systems with the development of the electro-mechanical system. NEC successfully delivered step-by-step, crossbar, and cross-reed technology to the world marketplace. During this era;

- In 1950 - NEC began manufacturing some of the world's first semiconductors
- In 1952 - NEC won the Deming prize awarded in communications
- In 1956 - NEC introduced step-by-step, cross-reed technology, and crossbar telephone switching systems to the world marketplace.
- In 1959 - NEC developed one of the first transistorized computers
- In 1963 - January 17, 1963 NEC entered the American marketplace with a sales office in New York
- In 1965 - NEC became a world innovator in digital transmission equipment

The Electronic Age

NEC entered the Electronic Age with solid-state technology, stored program control, and digital switching throughout the 1970's and 1980's.

- In 1972 - NEC developed the world's first 10K single element LSI chip
- In 1976 - NEC introduced the first skinny wire system
- In 1978 - NEC introduced the first digital hybrid, the Electra-100
- In 1979 - NEC entered the personal computer market
- In 1980 - NEC patented its Distributor Processor design (US Patent # 4,210,782). This little known patented process, invented by Kazunori Fujita, ushered in a new era of business communications systems through the use of distributed processing, modular building block concept, and fully integrated voice and data switching. Out of this patent came NEC's premier flagship product, the NEAX 2400 IMS
- In 1983 - NEC introduced the NEAX 2400 IMS
- In 1983 - NEC introduced one of the world's first super computers
- In 1985 - NEC introduced the Electra IMS
- In 1986 - NEC created one of the first 4MBit processor chips
- In 1988 - NEC opened the Software Development Center in the U.S. market
- In 1989 - NEC introduced the NEAX1400 IMS

The Information Age

Once again, NEC is delivering powerful communications systems. Systems that interface today's computers and telephones deliver a broad spectrum of integrated answers to today's modern communication challenges.

- In 1993 - NEC introduced the world's first 64M DRAM samples
- In 1993 - NEC introduced the NEAX 2400 ICS
- In 1994 - NEC introduced the NEAX 2000 IVS
- In 1997 - NEC introduced the NEAX 1000 IVS
- In 1998 - NEC introduced the NEAX 2400 IMX
- In 1999 - NEC introduced the NEAX 2000 IVS^2
- In 1999 - NEC introduced the NEAX EXPRESS
- In 2000 - NEC introduced the NEAX 2400 IPX
- In 2001 - NEC introduced the NEAX 2000 IPS
- In 2002 - NEC introduced the NEAX 2000 IPS DM
- In 2002 - NEC introduced the NEAX 2400 IPXi
- In 2003 - NEC introduced the NEAX 2000 IPS DMR
- In 2004 - NEC introduced the *UNIVERGE* SV7000
- In 2005 - NEC introduced the *UNIVERGE* NEAX 2000 IPS
- In 2005 - NEC introduced the *UNIVERGE* NEAX 2000 IPS DML
- In 2005 - NEC introduced the *UNIVERGE* NEAX 2400 IPX

NEC continues to focus its activities on the integration of computers and communications. NEC research and development, production, marketing, and service based on this integration positioned it to meet diversifying needs in worldwide markets.

Worldwide Leadership

NEC is committed to the linking people and information through technology, with over 15,000 different products distributed worldwide. Innovative, award-winning products featured in top publications form the foundation for NEC's leadership in these high-tech industries.

Components

NEC is the world leader in integrated components with billions of dollars in annual sales. NEC provides major processors and component chips to other leading manufacturers of communications, computers and electronic equipment.

- Very Large Scale Integrated (VLSI) memories
- Micro and Mini computers
- Application Specific Integrated Circuits (ASIC)
- Semiconductors
- A broad spectrum of other electronic components

Communications

NEC is one of the world's top suppliers in the production and distribution of communications equipment. With billions of dollars in sales, NEC is not the largest manufacturer of communications products, but offers the broadest spectrum of products and a demonstrated commitment to the total integration and networking of those products.

- Telephone company switching systems
- Network transmission systems
- Business communication systems
- Broadcast equipment
- Space electronics

Social Contributions

As a global corporation, NEC aims at achieving a higher level of coexistence with society. We will also strive to become an outstanding corporation respected by all those related to us and fully capable of offering high standards. In order to achieve this, we appreciate the great importance of contributing to society as a good corporate citizen and walk hand in hand with the world toward a better society.

Global Resources

The primary elements in a company's ability to respond to marketplace, buyer, and customer demands are the scope of its resources and strategy by which it uses those resources. NEC has established a worldwide network of human, physical, financial, and intellectual resources. These are deployed through localized operations, strategically located, professionally staffed, and thoroughly equipped. This enables NEC to deliver leading edge products and innovative services to businesses with hundreds of locations across the country or one location across town.

People

NEC has a global network of dedicated employees with offices in many countries. Our experienced management, engineers, installers, trainers, and staff pride themselves in a high level of job performance and customer support. More than 7,000 employees in the U.S. help maintain NEC's technological leadership with fresh ideas and the energy to turn them into reality.

Facilities

The physical resources of the company are the tangible assets that allow it to respond to your system's needs and your communication demands in a timely and effective manner. NEC takes pride in maintaining smooth working facilities with all resources required to provide the highest level of system service and client support. NEC has invested millions of dollars in facilities, tools, equipment, and inventory located in hundreds of corporate locations, direct offices, distributors, and affiliates worldwide to create one of the broadest multi-level support systems in the industry.

Assets

NEC's ability to manufacture and market its offerings depends in large part on its ability to remain profitable. With a substantial revenue base worldwide, NEC has built a strong financial and operational base -- a strong foundation of support our customers can depend on.

Innovation

Top training, knowledge, experience, and creativity are the hallmarks of intellectual excellence and of NEC's professional staff. Our telecom experts combine the skills and desire needed to provide the highest level of responsiveness to a system's service needs and your company's communication support requirements.

Community

NEC and its employees are active in worldwide, national, regional, and local community affairs through the dedication of their time and resources. NEC affiliates and their employees donate time, money, and equipment to help meet everyday and emergency needs of the healthcare and other social communities. In addition, the NEC businesses and their people give freely of their time, effort, and contributions to help meet the needs of local business groups, fraternal, and other social organizations.

Education

Through worldwide efforts, NEC soundly supports the education of citizens and nations around the world through financial grants, volunteer work, donations of equipment, and the sharing of knowledge. Specifically, the NEC Foundation of America is dedicated to encouraging and supporting the efforts of creative and effective non-profit organizations in the United States and their staffs, clients, and volunteers who, like NEC, work to develop and share the benefits of technology with society in order to help improve the quality of human life.

NEC Unified Solutions, Inc.

Established (PRIVATE)	1963, as a subsidiary of NEC Corporation, Tokyo, Japan
Headquarters	6555 North State Highway 161 Irving, Texas 75039-2402 PHONE: (214) 262-2000
Revenues	$3.7 billion
Employees	2,800+
Sales & Marketing Network	65 offices in 20 states throughout the United States, Puerto Rico and Canada.
Manufacturing Facilities	**HILLSBORO, OREGON** Fiber Optic Transport Systems Digital Loop Carrier Systems Very Small Aperture Terminals (Satellite) (VSATs) Asynchronous Transfer Mode (ATM) Switching Systems Digital PBXs Digital Multiplex Systems Automotive Electronic Systems Cellular Telephone and Pagers **TECNOLOGIAS NEC DE MEXICO, S.A. DE C.V.** Cellular Telephones and Pagers
Development Activities	**DALLAS, TEXAS** Advanced Switching Computer Telephony Software Radio & Satellite Communications Software Wireless Communication Products **HERNDON, VIRGINIA** Transmission Software & Firmware **SAN JOSE, CALIFORNIA** Transmission Software **HILLSBORO, OREGON** Transmission Hardware & Software

Major Product Areas

PUBLIC NETWORKING
Asynchronous Transfer Mode (ATM) Switching
Broadband Access Systems
Digital Loop Carrier Systems
Fiber Optic Transmission Systems
Microwave Radio Systems
Network Management Systems
Satellite Communications Systems
SONET Transport Systems
Internet Access Systems
Narrowband Wave Division Multiplexing Systems
Personal Access Communications System (PACS) Infrastructure Equipment

CORPORATE NETWORKING
Asynchronous Transfer Mode (ATM) Switches
Audio/Video Teleconference Systems
Telemedicine Products
Distance Learning Products
Data Communications Products
Key Telephone Systems
Local Area Network (LAN) interconnectivity
Private Branch Exchange (PBX) Systems, Software and Peripheral Products

COMMUNICATIONS TERMINALS
Analog & Digital Telephones
Cellular Mobile Telephones
Facsimile Equipment
Narrowband & Broadband Personal Communications Services (PCS) Products
Numeric & Alphanumeric Pagers
Enhanced Wireless Products (messaging, voice, data & accessories)

Scope Of Business

Development, manufacturing, sale, service and support of Telecommunications Products and Network Management Systems to the public and private sectors.

Engineering and Marketing of Integrated Computer and Communications (C&C) Systems and Networks including Broadband Multimedia Products.

Customization of ATM Switching Systems, Business Communication Systems, Data Communications, Radio and Transmission Equipment.

Vertical Markets

The *UNIVERGE* NEAX 2000 IPS is a full-featured IP based communications system that addresses the telecommunications market by providing a rich feature set of both Key and PBX features. The *UNIVERGE* NEAX 2000 IPS is positioned to provide a more cost effective solution for the small to medium size business, hotel property or networked environment. The *UNIVERGE* NEAX 2000 IPS provides unparalleled investment protection by allowing the user to take full advantage of today and tomorrow's technology advancements. The following is a list of some of the markets the IPS easily adapts to fit customer requirements.

- Business
- Telemarketing
- Communication
- Tenant Services
- Education
- Health Care
- Legal
- Hospitality
- Financial
- Government
- Transportation

Competitive Benefits and Advantages

- Advanced Technology (complete IP system on one card)
- Pentium Equivalent AMD CPU
- PSTN Gateway (requires trunk cards)
- Supports Both IP & TDM
- Both TDM & IP Totally Non-Blocking
- H.323 Connectivity
- Smaller Foot Print
- Low Power Consumption
- Wide Selection of IP and TDM Terminals
- Economical Incremental Licensing Costs
- Reduced Hardware Costs
- Survivability at the Remote Site
- Mobility and Continuance with in a Remote PIM Network

The *UNIVERGE* NEAX 2000 IPS provides a unique set of advantages to users who seek an advanced information system that is both flexible and dependable. Through the use of state-of-the-art computer controlled telecommunications technology, NEC is able to provide the following advantages:

1.) *Full-Featured System* – Station users have access to more than 300 service features that enhance user productivity, reduce operating costs, and improve communication efficiency. In addition, the control, network and interface positions can accommodate features, services and subsystems as required by specific applications.

2.) *Network Integration* - The *UNIVERGE* NEAX 2000 IPS offers business, industries, hospitals and hotel/motels the ability to access an extensive array of information processing and management services by serving as the central controller of an integrated information network. The *UNIVERGE* NEAX 2000 IPS reflects the philosophy of NEC Corporation to integrate C&C technology, and can provide many voice and non-voice services.

3.) *Flexible Line Size* - Innovative Modular hardware and software design allows from 64 ports up to 1020 ports in a main plus remote environment. Modularity gives the system the ability to expand from its minimum configuration to its maximum capacity as the need arises. This unique expansion capability allows the system to grow in a cost affective manner as the user requirements expand. Your initial system investment is protected through growth capability.

4.) *Energy Saving and Space Savings* - Through employment of state-of-the-art technology in the system circuitry design, NEC has been able to reduce power consumption. As a result, the current consumption of the system has been reduced to 50% of that of any conventional electronic PBX systems. This energy saving oriented system design allows for the use of much smaller capacity main power equipment and air conditioning equipment. In parallel with the energy savings, the space requirement for the system has been reduced to one third when compared with that of a conventional electronic PBX system.

5*.) Building Block Configuration* - In a conventional switching system, various kinds of equipment are mounted in a cabinet group and are connected to each other by use of connecting cables. The *UNIVERGE* NEAX 2000 IPS, however, uses a building block modular design. When installing the system, the required blocks are placed on top of each other in a building block formation and interconnected by round bus cables.

6.) *Flexible Interface Ports* - The *UNIVERGE* NEAX 2000 IPS employs a Universal Port architecture that has the flexibility to accommodate station terminal equipment, trunks and adjunct processors. This universality allows the *UNIVERGE* NEAX 2000 IPS to optimize use of slot space and lower expansion costs.

7.) *High Reliability* - The *UNIVERGE* NEAX 2000 IPS is designed and manufactured to provide the highest level of system reliability. The *UNIVERGE* NEAX 2000 IPS is designed with such features as: remote maintenance, distributed call processing, error-correcting memory, battery backup, and automatic system alarm indications to insure unsurpassed reliability. Only the finest components have been used. In addition, through the employment of LSI and custom LSI and VLSI technology, the number of component parts has been greatly reduced, thus lessening possible failures and insuring continuous operation.

8.) *Intelligent Attendant Console* - The *UNIVERGE* NEAX 2000 IPS attendant console, SN716 Desk Console, is a compact, desktop unit equipped with non-locking keys, Light Emitting Diodes, and a 4 x 40 character Liquid Crystal Display. The LED's provide continuous information relative to the status of calls in progress. The display provides station and trunk identification, class of service and the number of calls waiting. The LCD changes with different call states and instructs the user which Multifunction keys are available for each state. Keys such as Busy Verify, DND Override, etc. only appear when needed. The Multi-function Keys reduce the number of different buttons and greatly simply operation.

9.) *Intelligent Digital Multifunction Terminal* - In addition to supporting conventional station equipment, the *UNIVERGE* NEAX 2000 IPS can be equipped with the Dterm series digital electronic multifunction terminals. The Dterm terminals are intelligent microprocessor controlled terminals, which enhance the feature capabilities offered by the system and provide the service of conventional key telephones over 1-pair wiring. The Dterm instrument may be equipped with an interface adapter to allow simultaneous voice and data switching, without compromising the voice communication system.

10.) *Ease of Installation* - Because the *UNIVERGE* NEAX 2000 IPS uses pre-assembled modules and plug-in type circuits packs, it is easy to install. Wiring connections, both internal and external, are made through simple-to-use standard plug-ended cables. In addition, with each unit and system having been fully factory tested prior to shipment, potential obstacles to easy installation have been held to a minimum.

11.) *Ease of Maintenance* - Because the system is constructed with first quality components, reliability is high and operation is trouble free. However, if a minor fault occurs, the self-diagnostic programs will detect the fault, and automatically make the needed corrections. If the problem is beyond the internal correction capabilities of the system, the self-diagnostic programs will automatically print the nature of the fault and the involved unit is identified on the man-machine interface equipment. The faulty plug-in unit can then be quickly replaced with little or no interruption of service.

12.) *Flexible Numbering Plan* - The *UNIVERGE* NEAX 2000 IPS provides flexible numbering assignment to meet all forms of network integration service.

13.) *Future Capabilities* - Because the *UNIVERGE* NEAX 2000 IPS uses a stored program control, performance enhancements and new features can be easily incorporated by simple changes in software. The *UNIVERGE* NEAX 2000 IPS can be upgraded by using the most current software release and, when necessary, additional hardware. This means that the system will not become obsolete.

14.) *Cost Controls* - With telecommunications costs growing, it is becoming increasingly important to control them. The *UNIVERGE* NEAX 2000 IPS makes it possible for you to get a firm grasp on telecommunications costs. Through the use of such features as least cost routing, class of service and detail call recording, cost reduction and control are possible.

UNIVERGE NEAX 2000 IPS

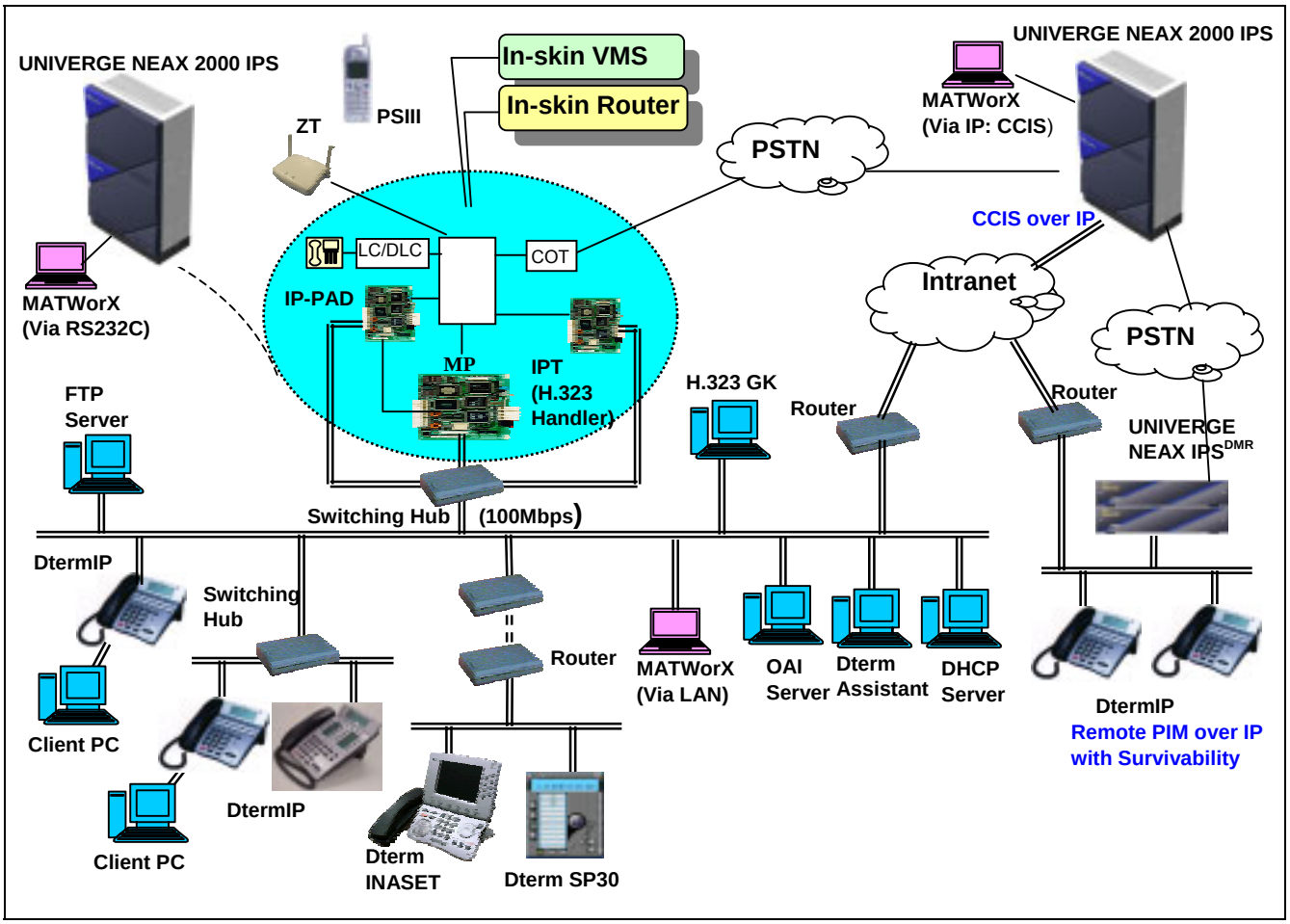
In the modern information-based economy, timely access to accurate information is an essential resource for every successful organization. In many cases it is the primary commodity being created, bought, or sold and anything less than the best communication system puts your organization at a disadvantage. The complex interconnections of voice, print, and data communications demand a powerful but elegant system capable of taking you through the 21st century.

The *UNIVERGE* NEAX 2000 IPS (Internet Protocol Server) is a full-featured IP based communications system providing a rich feature set with pure Voice over IP (VoIP) communications (peer to peer connections), across corporate Local and Wide Area Networks (LAN and WAN). DtermIP telephones are designed to provide a converged infrastructure at the desktop, with a 100 Base T Ethernet connection to the LAN and built-in hub for a PC connection to the telephone itself. The system can provide peer-to-peer connections between DtermIP telephones with voice compression, offering existing Dterm Series i telephone features. On the WAN side, the system can provide peer-to-peer connections over IP networks with the voice compression, on a CCIS basis (CCIS over IP) or Remote PIM (Remote PIM over IP).

The *UNIVERGE* NEAX 2000 IPS can provide legacy station/trunk interfaces to support the existing Time Division Multiplexing (TDM) based infrastructure, such as analog telephones, analog networks, and digital networks (T1/E1, ISDN etc.). At maximum configuration, the system can provide 1020 ports for IP and legacy devices, and 256 ports for Application cards. Communications between legacy stations/trunks and DtermIP telephones/IP networks are made via IP PAD, which converts packet-based voice data to TDM-based voice data, and vice versa. Both peer-to-peer connections and TDM-based connections are controlled by the Main Processor (MP) card. The MP card incorporates a built-in Device Registration Server (DRS) and a single interface point of IP connection to IP telephone, MATWorX, Voice Mail and OAI/ACD servers.

UNIVERGE NEAX 2000 IPS users have access to hundreds of service features that are used in building unique telephony applications that enhance productivity, reduce operating costs and improve communications efficiently. The innovative modular hardware and software design allows efficient, effective growth within each module from its minimum to its maximum configuration. The *UNIVERGE* NEAX 2000 IPS software design is as advanced as its hardware. It ensures the system will support evolving applications and have the reliability needed to compete in today's world and into tomorrow's. The software is designed with modularity in mind. Together, these modular building blocks allow customers to initially buy what they need and add capacity and capabilities as the business demands, resulting in a greater degree of cost control for new installations and for upgrades to features, capacities and the software versions.

The illustration below shows a typical system layout.



Station to Station Connection

For DtermIP to DtermIP connection (Peer to Peer connection), the voice data is transmitted and received directly between DtermIPs on the LAN. For Dterm Legacy terminal connection, the IP-PAD card and VCT card are required to transmit and receive the voice data. These cards are used to control and convert the voice data. The MP card in either of the connections above manages the control signals.

CCIS Connection

DtermIP to DtermIP connection (Peer to Peer connection) via CCIS is available only when the destination office is *UNIVERGE* NEAX 2000 IPS or *UNIVERGE* NEAX 2400 IPX. The system provides only Point to Multipoint connection.

Maintenance

MATWorX IPS is used as the maintenance program for the *UNIVERGE* NEAX 2000 IPS. Direct connection (RS-232C), Modem connection and LAN (TCP/IP) connections are available to connect to the MAT (Maintenance Administration Terminal).

Hybrid System of IP (peer-to-peer connection) and TDM Switching

The *UNIVERGE* NEAX 2000 IPS supports both pure IP switching (peer-to-peer connections) and Time Division Switching (TDM). The pure IP switching is provided for communications between DtermIPs and for CCIS/Remote PIM connections with another *UNIVERGE* NEAX 2000 IPS/IPS DM/IPS DML/2400 IPX (CCIS over IP or Remote PIM over IP). On the other hand, the TDM switching is provided for communications between legacy stations/trunks. Connections between DtermIP/CCIS or Remote PIM over IP and legacy stations/trunks are made via IP PADs, which converts packet-based voice data to TDM-based voice data, and vice versa.

Built-in DRS (Device Registration Server) on MP

The *UNIVERGE* NEAX 2000 IPS incorporates DRS (Device Registration Server) on MP, which provides a Log-in/Log-out management of DtermIP including Registration, Authentication. Also the built-in DRS can be inter-worked with DHCP server to provide easy administration of IP address.

Reduced Hardware with IP based Architecture

The DtermIPs accommodated in a LAN do not require a DLC card because they can be interfaced directly with the LAN and connected on a peer-to-peer basis. When a DtermIP is connected with station/trunk, which is accommodated in TSW, the speech path between LAN and TSW is made via IP PAD under the call processing control of MP. The number of DtermIPs can be simply expanded by adding the terminal itself, an IP Seat License and IP PADs if traffic volume is increased. With this system architecture, the hardware such as DLC, PIM, Power Supply etc. is reduced and easy cost effective move, add, change is realized.

Office Data Backup Enhancement

The office data of *UNIVERGE* NEAX 2000 IPS is stored in Flash ROM.

Various Installation Methods

To meet the specific needs of the customers' environment, the *UNIVERGE* NEAX 2000 IPS provides the following installation methods:

Floor Standing Installation
Wall-mounting Installation
IEC standard 19-inch Rack-mounting Installation

Unified Circuit Card Size

All the circuit cards for *UNIVERGE* NEAX 2000 IPS are designed in one size (PN-type), and installed in the PIM. This maximizes the efficiency of slot utilization of the PIM.

High Density Line/Trunk Cards

Major line/trunk cards used in *UNIVERGE* NEAX 2000 IPS are provided with 8 circuits per card. This allows the physical system size to be compact.

Universal Slot

One PIM provides 12 card slots for Line/Trunk (LT). Also, these card slots can be used for Application Processor (AP) cards without complicated limitations. As this makes easy quotation and installation, more number of AP cards can be mounted in one PIM.

DC/DC Power Supply for –48V

The PIM houses optional DC/DC Power Supply for the cards which require –48V power such as CSI card used for interface of Cell Station (CS) of wireless system. Since this power supply is mounted in the space under the AC/DC power, additional Power Module/card slots are not required.

Extended Application Processor (AP) Port Capacity

The *UNIVERGE* NEAX 2000 IPS provides a maximum of 256 AP ports and it is independent of the 512 physical ports for legacy Line/Trunk (LT), therefore, more AP cards can be used in the system such as T1/E1 digital link, etc.

Dual MP System

The system complies with dual control system on Main Processor.

Note: *Since the system employs Cold Standby processing in MP changeover, the calls in progress are terminated as a result of the MP changeover. Also, during the MP changeover, the call originating/receiving and service feature access are not effective. (It takes about 30 to 60 seconds to complete the MP changeover.)*

UNIVERGE NEAX IPS DM

The *UNIVERGE* NEAX IPS DM (Internet Protocol Server Distributed Model) is equipped with all the features and functions of the *UNIVERGE* NEAX 2000 IPS, with a smaller space requirement. It is a full-featured PBX that supports advanced networking, pure peer-to-peer IP telephony connectivity and traditional TDM switching capabilities. Designed primarily for pure converged IP networks, the *UNIVERGE* NEAX IPS DM can also accommodate a mixed (i.e., TDM and IP) converged IP network or standalone solution.

The *UNIVERGE* NEAX IPS DM supports up to 952 peer-to-peer IP stations and 56 TDM ports in a single modular chassis. Up to two chassis can be stacked providing maximum capacity of 112 legacy TDM ports while still supporting as many as 888 peer-to-peer IP stations or more depending on the amount of TDM stations used. It uses the same CPU, line/trunk cards, application processor cards and software of the *UNIVERGE* NEAX 2000 IPS and comes equipped for 19” rack mounting. It offers superior port density; each chassis only occupies two Rack Units (2RU).

UNIVERGE NEAX IPS DML

The *UNIVERGE* NEAX IPS DML (Internet Protocol Server Distributed Model Local) is a *UNIVERGE* NEAX IPS DM that has been optimized for Small Office Stand Alone IP Solution with from 10 to 100 IP telephones. The *UNIVERGE* NEAX IPS DML uses the SPN-CP31C as the Main Processor. The SPN-CP31C is a cost down CPU to compete with Mitel 3100, NBX25, and CISCO 2600 Series. This system targets users with up to 112 TDM station, 128 IP stations and can be used as an end point in a peer-to-peer CCIS network. The following options that are built-in on the CP24 are not available with the CP31:

- No built-in DAT.
- Only one RS Port.
- No built-in DK (external/relay key).
- No MN Alarm Indication

UNIVERGE NEAX IPS DMR

The *UNIVERGE* NEAX IPS DMR (Internet Protocol Server Distributed Model Remote) is an *UNIVERGE* NEAX IPS DM that has been optimized for Remote PIM over IP applications. The *UNIVERGE* NEAX IPS DMR uses the SPN-CP31C as the Main Processor. The SPN-CP31C is a cost down CPU to compete with Mitel 3100, NBX25, and CISCO 2600 Series. This system targets users who have up to 30 relatively small offices that accommodate 10-30 extensions at the Remote Site. The MP card at Remote Site has the same system data as that at Main Site, because Remote Site automatically gets the data from Main Site at the time of setup. In normal operation, Main Site automatically copies the system data to Remote Site through the network once a day.

The following options that are built-in on the CP24 are not available with the CP31 in an *UNIVERGE* NEAX IPS DMR:

- No built-in DAT.
- No built-in Modem
- Only one RS Port.
- No built-in DK (external/relay key).
- No MN Alarm Indication

System Outline for Remote PIM

- The MP card at Main Site controls system processing, and Remote Site follows the Main Site.
- Remote Site can accommodate most terminals and trunks such as Dterm, Single-Line telephone, DSS/BLF Console, DtermIP, COT, ISDN, etc. The Attendant Console, Dterm Attendant position, and Add-on Module are **not** supported at the Remote Site.
- Local Switch (TDSW) at Remote Site controls connections within the Remote Site if possible.
- In the case of connections between Main-Remote and Remote-Remote, the voice path is connected via Peer-to-Peer or IP-PAD.
- If the communications between Main-Remote are interrupted, the Remote Site survives by itself after a system reset.

Advantages of Remote PIM

- The system regards the terminals accommodated in both Main Site and Remote Site as the extensions in the same office. Therefore, the service transparency is superior to CCIS.
- Remote PIM over IP has no limitation of distance between Main and Remote.
- Remote Site has a switching function at local. This provides the effective configuration of C.O. line. In addition, the Remote Site can accommodate AP cards. This is an advantage to accommodate ISDN lines especially.
- The Remote Site survives by itself even if the link between Main and Remote is disconnected. Therefore, the impact to users at the Remote Site will be smaller if the link between Main and Remote is disconnected.
- This feature can reduce the bandwidth used on the WAN that is connected to CO lines at Remote Site, rather than DtermIP at remote location or the Media Converter (MC) accommodation.

Characteristics of the *UNIVERGE* NEAX IPS DM/IPS DML/IPS DMR

Compact and Small Size MODULAR CHASSIS

One MODULAR CHASSIS provides 8 card slots /56LT ports (8 virtual LT ports are available per MODULAR CHASSIS in addition to 56LT ports).

- 2 MODULAR CHASSIS can be used per system for IPS DM/DML/IPS DMR

MP (Main Processor)

- PN-CP24A/B/C for IPS DM, the same MP as the *UNIVERGE* NEAX2000 IPS.
- PN-CP31C for IPS DML and PN-CP31A/C for IPS DMR

Power Failure Transfer (PFT)

Power Failure Transfer (PFT) for the IPS DM is provided with PZ-4PFTA card. The PZ-8PFTB is not available for the IPS DM.

Installation Methods

The *UNIVERGE* NEAX IPS DM/IPS DML/IPS DMR can be installed on a desktop or into the 19-inch rack. Wall Mount Installation is not available.

Remote PIM over IP with Survivability

The *UNIVERGE* NEAX 2000 IPS can have a PIM installed at a remote site through an IP network. At the main site, the *UNIVERGE* NEAX IPS/IPS DM is installed and *UNIVERGE* NEAX 2000 IPS/IPS DM/IPS DMR or IPS Retro is installed at the remote site. The main site controls call processing and service feature access for station users located at both the main and remote sites. When the Remote PIM cannot be connected with main site due to the IP network and/or main PBX failure, the Remote PIM initializes the system and re-starts operation by its own Main Processor (survival mode). In the survival mode, almost all service features are provided to the station users accommodated in Remote PIM. When the IP network/main PBX recovers, the Remote PIM can be restored to normal mode with a system initialization by manual operation or automatically (Selectable by system data setting).

- IPS DMR with CP31A/C MP
- IPS PIM with CP24A/B/C or CP27A MP
- IPS Retro with CP26 or CP28 MP

Stand Alone System Capacity

IPS System Capacity (Single MP System)

Item		Capacity Per PIM Note							
		PIM1	PIM2	PIM3	PIM4	PIM5	PIM6	PIM7	PIM8
LT Card	(No. of Ports)	64	128	192	256	320	384	448	512
	(No. of Cards)	12	24	36	48	60	72	84	96
AP Card	(No. of Ports)	Max. 256 ports per system							
	(No. of Cards)	12	24						
Total Number of Lines (Single Line Tel. + Dterm)		64	128	192	256	320	384	448	512
IP PAD	(No. of Channel)	64		128		192		256	
Single Line Telephone (Lines)	Standard	64	128	192	256	320	384	448	512
	Long	48	96	144	192	240	288	336	384
Dterm (Lines)	Standard	64	128	192	256	320	384	448	512
	Long	24	48	72	96	120	144	168	192
Dterm IP/Dterm IP INASET (PTP Connection)		952	888	824	760	696	632	568	504
Dterm PS		512							
Cell Station (CS) / Zone Transceiver (ZT)		16	32	48	64	80	96	112	128
ISDN Station		16	32	48	64	80	96	112	128
Central Office Trunk (Lines)	Loop Start	64	128	192	256	256	256	256	256
	DID w/4DID	48	96	144	192	240	256	256	256
Tie Line Trunk (Lines)	2W E&M	24	48	72	96	120	144	168	192
	4W E&M	24	48	72	96	120	144	168	192
CCIS Trunk (Peer to Peer Connection)		Max. 127							
DTI/CCIS Digital Link	1.5M	DTI: 10, CCIS: 8							
	2MI	8							
ISDN	1.5M/2M (PRT)	8							
	4BRT (card)	6	12	18	24	24	24	24	24
IP Trunk		1	2	3	4	5	6	7	8
PFT Connections		8	16	24	32	40	48	56	64
3-Party Conference		Max. 16 conference groups per system							
6-/10-Party Conference	6-Party	Max. 4 conference groups per system							
	10-Party	Max. 2 conference groups per system							
32-Party Conference		Max. 8 conference group per system							
Built-in Router		Max. 8 cards per PIM							
DTMF Sender		Max. 32 circuits per system							
DTMF Receiver		16		32					
Attendant Consoles		8							

IPS System Capacity (Single MP System cont'd)

Item	Capacity Per PIM Note							
	PIM1	PIM2	PIM3	PIM4	PIM5	PIM6	PIM7	PIM8
Attendant Terminal (Dterm ATT Position)	Max. 8 sets per system							
SMDR Interface	Max. 2 Interface ports (RS232) Max 1 interface port (IP) per system							
PMS Interface	Max. 1 Interface port (RS232) Max 1 interface port (IP) per system							
ACD / MIS or OAI Interface	Max. 1 interface port per system							
Remote PIM over IP (Number of PIM for Remote Sites)	Up to 30 (depending on network)							
DID Dial Conversion	1000							
Call Forwarding-Outside Set	496							
Authorization. Code / Forced Account Code / Remote Access to System(DISA)Code	3000							
Message Reminder Set	1024							
Name Display / Guest Name Display	512							
Speed Calling-Station (Station Speed Dial) Set	10000							
MP built-in SMDR Call Record	1024							

Note: *System Capacity is for Main site only. For Total System Capacity see IP Remote Network System Capacity.*

IPS System Capacity (Dual MP System)

Item		Capacity Per PIM Note							
		PIM1	PIM2	PIM3	PIM4	PIM5	PIM6	PIM7	PIM8
LT Card	(No. of Ports)	64	128	192	256	320	384	448	512
	(No. of Cards)	11	23	35	47	59	71	83	95
AP Card	(No. of Ports)	Max. 256 ports per system							
	(No. of Cards)	11	23	24					
Total Number of Lines (Single Line Tel. + Dterm)		64	128	192	256	320	384	448	512
IP PAD (No. of Channel)		64		128		192		256	
Single Line Telephone (Lines)	Standard	64	128	192	256	320	384	448	512
	Long	44	92	140	188	236	284	332	380
Dterm (Lines)	Standard	64	128	192	256	320	384	448	512
	Long	22	46	70	94	118	142	166	190
Dterm IP/Dterm IP INASET (PTP Connection)		952	888	824	760	696	632	568	504
Dterm PS		512							
Cell Station (CS) / Zone Transceiver (ZT)		16	32	48	64	80	96	112	128
ISDN Station		16	32	48	64	80	96	112	128
Central Office Trunk (Lines)	Loop Start	64	128	192	256	256	256	256	256
	DID w/4DID	44	92	140	188	236	256	256	256
Tie Line Trunk (Lines)	2W E&M	22	46	70	94	118	142	166	190
	4W E&M	22	46	70	94	118	142	166	190
CCIS Trunk (Peer to Peer Connection)		Max. 127							
DTI/CCIS Digital Link	1.5M-AMI	DTI: 10, CCIS: 8							
	2M-AMI	8							
ISDN	1.5M/2M (PRT)	8							
	2BRT (card)	11	23	24					
	4BRT (card)	6	12	18	24	24	24	24	24
IP Trunk		1	2	3	4	5	6	7	8
PFT Connections		8	16	24	32	40	48	56	64
3-Party Conference		Max. 16 conference groups per system							
6-/10-Party Conference	6-Party	Max. 4 conference groups per system							
	10-Party	Max. 2 conference groups per system							
32-Party Conference		Max. 8 conference group per system							
Built-in Router		Max. 8 cards per PIM							
DTMF Sender		Max. 32 circuits per system							
DTMF Receiver		16		32					
Attendant Consoles		8							

IPS System Capacity (Dual MP System, Cont'd)

Item	Capacity Per PIM Note							
	PIM1	PIM2	PIM3	PIM4	PIM5	PIM6	PIM7	PIM8
Attendant Terminal (Dterm ATT Position)	Max. 8 sets per system							
SMDR Interface	Max. 2 Interface ports (RS232) Max 1 interface port (IP) per system							
PMS Interface	Max. 1 Interface port (RS232) Max 1 interface port (IP) per system							
ACD / MIS or OAI Interface	Max. 1 interface port per system							
Remote PIM over IP (Number of PIM for Remote Sites)	Up to 30 (depending on network)							
DID Dial Conversion	1000							
Call Forwarding-Outside Set	496							
Authorization. Code / Forced Account Code / Remote Access to System(DISA)Code	3000							
Message Reminder Set	1024							
Name Display / Guest Name Display	512							
Speed Calling-Station (Station Speed Dial) Set	10000							
MP built-in SMDR Call Record	1024							

Note: *Capacity is for Main site only. For Total System Capacity see IP Remote Network System Capacity.*

IPS DM/IPS DML System Capacity

Number of PHYSICAL MODULAR CHASSIS		Capacity Per MC	
		1	**2**
LT card **Note 1**	No. of ports	56	112
	No. of cards	7	14
AP card	No. of ports	Max. 256 ports per system	
	No. of cards	7	14
Total number of lines (Single Line Telephone + Dterm)		56	112
IP-PAD	No. of channel	32	64
Single Line Telephone (Lines)	8LC	56	112
	Long Line	Not Available	
Dterm(Lines)	Standard	56	112
	Long Line	8	16
DtermIP/DtermIP/INASET/SP30(PTPConnection)**Note 3**		952	888
ISDN Station		10	20
Central Office Trunk (Lines)	Loop Start	56	112
	DID w/4DID	28	56
	2W/4W E&M	14	28
CCIS Trunk (Peer to Peer Connection)		Max. 127	
DTI/CCIS Digital Link **Note 2**	1.5M	7	DTI: 10, CCIS: 8
	2M	7	8
ISDN	1.5M/2M(PRT)	7	8
	4BRT (card)	5	10
IP Trunk		1	2
PFT Connections		4	8
3-Party Conference		Max. 16 conference groups per system	
6-/10-Party Conference	6-Party	Max. 4 conference groups per system	
	10-Party	Max. 2 conference groups per system	
32-Party Conference		5	Max. 8 conference groups per system

IPS DM/IPS DML System Capacity Cont

Number of PHYSICAL MODULAR CHASSIS	Capacity Per MC	
	1	**2**
Built-in Router	Max. 1 card per Site	
DTMF Sender	Max. 32 circuits per system	
DTMF Receiver	16	32
SN716 Desk Console	8	
Attendant Terminal (Dterm ATT Position)	Max. 8 per system	
SMDR Interface	Max. 2 Interface ports (RS232) Max 1 interface port (IP) per system	
PMS Interface	Max. 1 Interface port (RS232) Max 1 interface port (IP) per system	
ACD / MIS or OAI Interface	Max. 1 Interface port per system	
Remote PIM over IP (Number of PIM for Remote Sites) **Note 3**	Up to 30 (depending on network)	
DID Dial Conversion	1000	
Call Forwarding-Outside Set	496	
Authorization Code / Forced Account Code / Remote Access to System(DISA) Code	3000	
Message Reminder Set	1024	
Name Display / Guest Name Display	512	
Speed Calling-Station (Station Speed Dial) Set	10000	
MP built-in SMDR Call Record **Note 3**	1024	

Note 1: *Each Modular Chassis has 8 Virtual LT Ports that can only be used to expand the PAD channels from 24 to 32 using the 8IPLA w/24IPLA.*

Note 2: *The total number of trunk line and DTI channel shall be 256 or less. (Each trunk line and DTI channels are required to assign the "Trunk Number" by system data programming and maximum number of system parameter for "Trunk Number" is 256.)*

Note 3: *IPS DML only supports a maximum of 128 IP stations; IPS DML does not support built-in SMDR. The IPS DML is a Stand Alone Only solution; Remote PIM's off the IPS DML are not supported.*

IP Remote Network Capacity

Total System Capacity (Main plus Remote)

Item		Capacity
LT Ports		1020
AP Ports		256
Analog Single Line Tel. + Dterm		980
IP PAD (No. of Channel)		256
Dterm IP/Dterm IP INASET (PTP Connection)		952
Dterm PS		512
Cell Station (CS) / Zone Transceiver (ZT)		128
ISDN Station		128
Central Office Trunk (Lines)		256
Tie Line Trunk (Lines)	2W/4W E&M	192
CCIS Trunk (Peer to Peer Connection)		127
DTI/CCIS Digital Link	1.5M/2M	DTI: 10/CCIS: 8 Links
ISDN	1.5M/2M (PRT)	8
	4BRT (card)	24
IP Trunk		8
PFT Connections		64
3-Party Conference		Max. 16 conference groups
6-/10-Party Conference	6-Party	Max. 4 conference groups
	10-Party	Max. 2 conference groups
32-Party Conference		Max. 8 conference groups
Built-in Router		1 per Site
DTMF Sender/Receiver		Max. 32 circuits
Attendant Consoles		8
Attendant Terminal (Dterm ATT Position)		Max. 8 sets
SMDR Interface		Max. 2 Interface ports (RS232) Max 1 interface port (IP) per system
PMS Interface		Max. 1 Interface port (RS232) Max 1 interface port (IP) per system
ACD / MIS or OAI Interface		Max. 1 interface port
Remote PIM over IP **Note**		Up to 30 (depending on network)
DID Dial Conversion		1000
Call Forwarding-Outside Set		496
Authorization Code / Forced Account Code / Remote Access to System(DISA)Code		3000
Message Reminder Set		1024
Name Display / Guest Name Display		512
Speed Calling-Station (Station Speed Dial) Set		10000
MP built-in SMDR Call Record		1024

Note: *Remote PIM networks using the IPS DML as the main are not supported.*

IPS DMR Capacity

Number of PHYSICAL MODULAR CHASSIS		Capacity Per MC	
		1	2
LT card **Note 1**	No. of ports	56	112
	No. of cards	7	14
AP card	No. of ports	Max. 256 ports per network	
	No. of cards	7	14
IP-PAD	No. of channel	32	64
Single Line Telephone (Lines)	8LC	56	112
Dterm(Lines)	Standard	56	112
	Long Line	8	16
DtermIP/DtermIP/INASET(Peer to Peer Connection) **Note 2**		128	
Central Office Trunk (Lines)	Loop Start	56	112
	DID w/4DID	28	56
	2W/4W E&M	14	28
DTI	1.5M	7	10
ISDN	1.5M(PRT)	7	8
	4BRT (card)	5	10
PFT Connections		4	8

IPS PIMMJ (As Remote PIM) Capacity

Number of PHYSICAL PIMS		Capacity Per PIM	
		1	2
LT card	No. of ports	64	128
	No. of cards	8	16
AP card	No. of ports	Max. 256 ports per network	
	No. of cards	12	24
IP-PAD	No. of channel	32	64
Single Line Telephone (Lines)	8LC	64	128
Dterm(Lines)	Standard	64	128
	Long Line	24	48
DtermIP/DtermIP/INASET(Peer to Peer Connection) **Note 2**		128	
Central Office Trunk (Lines)	Loop Start	64	128
	DID w/4DID	48	96
	2W/4W E&M	24	48
DTI	1.5M	10	
ISDN	1.5M(PRT)	8	
	4BRT (card)	6	12
PFT Connections		8	16

Note 1: *Each Modular Chassis has 24 Virtual LT Ports that can only be used to expand the PAD channels from 8 to 32 using the 8IPLA w/24IPLA.*

Note 2: *Remote PIM's Support up to 2 Virtual PIM's for assignment of DtermIP/DtermIP/INASET only.*

Chapter 2 System Architecture

The *UNIVERGE* NEAX 2000 IPS family consists of the IPS, IPS DM, IPS DML and IPS DMR. These *UNIVERGE* NEAX 2000 Internet Protocol Servers fuse existing NEC technologies with dynamic advancements in hardware and software to satisfy the most stringent system requirements. NEC's modular design, sensible migration, comprehensive network solutions, and the ability to customize functionality to match specific business applications are just a few of the NEC benchmarks that the new IPS builds upon.

UNIVERGE NEAX 2000 IPS users have access to hundreds of service features that enhance productivity, reduce operating costs, and improve communications efficiently. The innovative modular hardware and software design allows efficient, effective growth within each module from its minimum to its maximum configuration. This is achieved by a unique building block architecture allowing growth without the loss of existing hardware or software.

Voice and Data Switching

The *UNIVERGE* NEAX 2000 IPS systems support advanced applications such as Tandem Networking, Centralized Attendant Service, Direct Digital Interface and other voice features. The system's non-blocking architecture and distributed processor control hierarchy are designed to support the traffic and control the load generated by voice and data switching. Station users may perform simultaneous voice and data transmission at speeds up to 19.2Kbps over universal 1-pair wiring without the use of modems. Proprietary Digital Instruments (Dterm) may be provided to increase system flexibility and eliminate the need for conventional multiple line stations with their associated control equipment and cable plant requirements.

Hardware Architecture

The *UNIVERGE* NEAX 2000 IPS systems employ the latest advancements in Large Scale Integration (LSI) circuits and component manufacturing techniques to create a highly reliable and serviceable communications system. The IPS has changed from NEC's traditional PROM based CPUs to a Flash ROM based for system software and RAM for data programming. The *UNIVERGE* NEAX 2000 IPS architecture consists of three major functional components: Distributed Controller, Digital Switching Network, and Port Interface.

1. Distributed Controller - The Distributed Controller is composed of distributed multiprocessing units, generic memory, database instructions, system interface, and interface ports for system maintenance and administration.

2. Digital Switching Network - The Digital Switching Network consists of a non-blocking digital time division switch, allowing all ports to be used simultaneously.

3. Port Interface - The Port Interface provides access to the public and private network for various types of terminal devices, including digital and analog telephones, data terminals, computers and subsystems such as Voice Mail Systems, Data Switch Networks, and related communication and information services.

Hardware Design

The *UNIVERGE* NEAX 2000 IPS hardware is designed to provide the best of the following:

- Flexibility
- Capacity
- Reliability
- Optimum use of space
- Minimal environmental requirements
- Ease of installation and maintenance

A unique modular design is employed throughout the *UNIVERGE* NEAX 2000 IPS line of products. As additional equipment modules are required, they are stacked on top of each other, rather than mounting the equipment in conventional frames or cabinets. This innovative method reduces installation time, avoids the use of bulky frames and provides for manageable future expansion, virtually eliminating the possibility of outgrowing your *UNIVERGE* NEAX 2000 IPS.

Fusion of RAM and ROM Programs

The *UNIVERGE* NEAX 2000 IPS incorporates the flexibility of Random Access Memory (RAM) for Office Data Memory and Flash ROM that contains the operating system. This fusion of divergent memory technologies results in high performance 32-bit processing reliability in system operation as well as fast recovery time in case of system power loss. This unique configuration provides improved efficiency for updates to the generic program by floppy disk, and simplifies upgrades for system features and capabilities throughout its life cycle.

Main Processor (MP) with Integrated Functionality

The *UNIVERGE* NEAX 2000 IPS Main Processor (MP) is a heart of pure IP connections and TDM-based connections. The MP employs a high-speed CPU, which is equivalent with Pentium. With this processing power and System On Chip (SOC) technology, the MP integrates Device Registration Server (DRS), AP01 (OAI) functions*, which are provided by an additional card in the previous IVS^2. Also, by means of today's advanced LSI technology, the MP card size is minimized and On-board Ethernet Interface Card is mounted on MP without using an additional slot space in PIM. This interface card is linked with LAN for call control processing of DtermIP and inter-work with MATWorX and OAI server. (*RS-232C interface, Authorization Code service and FLF facility are deleted.)

Application Processors

The processing architecture of the *UNIVERGE* NEAX 2000 IPS allows the system to be configured with Application Processors that provide processing capacity for a specific application or multiple applications. These EPROM-based processors support applications such as:

- ACD Processor for call center applications
- SMDR for call accounting
- Hotel Processor for Hospitality applications
- Message Center Interface (MCI)
- Open Applications Interface (OAI), which connects to external computing devices to support a wide range of host computer-supported features.

These application processors provide the power and flexibility to meet specific users' applications and traffic needs.

Integration

In addition to the application processors manufactured by NEC, the *UNIVERGE* NEAX 2000 IPS systems also deliver an open architecture for integration with other manufacturers. Interface processors and specifications for communicating with computing devices support this integration. These computing devices enable such service features as network and facilities management, call center applications, call accounting, property management, voice mail, maintenance access terminals, and management information systems for ACD reports and monitoring.

Unified Circuit Card

All the circuit cards for *UNIVERGE* NEAX 2000 IPS are designed in one size (PN-type), and installed in the PIM. This maximizes the efficiency of slot utilization of the PIM.

High Density Line/Trunk Cards

Major line/trunk cards used in *UNIVERGE* NEAX 2000 IPS are provided with 8 circuits per card. This allows the physical system size to be compact.

Universal Slot

One PIM provides 12 card slots for Line/Trunk (LT). Also, these card slots can be used for Application Processor (AP) cards without complicated limitation. As this makes easy quotation and installation, more number of AP cards can be mounted in one PIM.

DC/DC Power Supply for –48V

The PIM houses an optional DC/DC Power Supply for the cards which require –48V power such as the CSI card used for interface of Cell Station (CS) of a wireless system. Since this power supply is mounted in the space under the AC/DC power, additional Power Module/card slots are not required.

SN716 Attendant Console

The SN716 Attendant Console connection is available through one twisted-pair cable when using an AC adapter. When using the PN-PW00, one twisted-pair cable is required for voice/data and one pair is required for power from the PW00 to the Console.

Analog/Digital Telephones

The analog/digital telephone connection is available through one twisted-pair cable.

IP Telephones

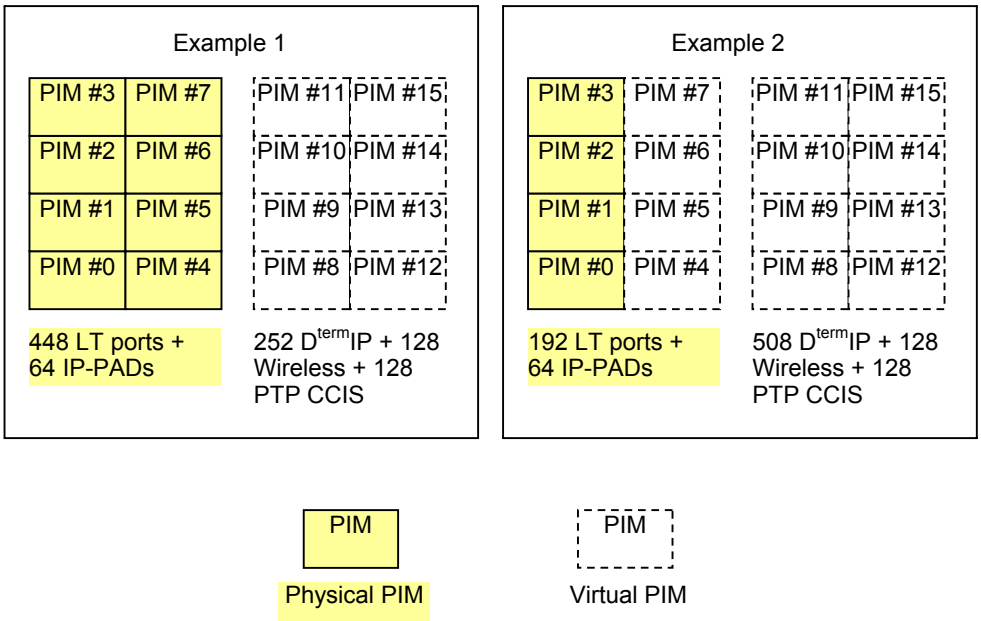
The IP telephones can communicate with other IP telephones over the LAN, on a peer-to-peer connection basis. The Main Processor (MP) provides call control with an Ethernet adapter card. Voice packets are transmitted between IP telephones over the LAN (not through Time Division Switch). Voice compression of G.729a (8 kbps) and G.723.1 (5.3 kbps/6.3 kbps) is available for these connections.

The IP telephones can communicate with legacy stations and trunks (including H.323 trunks) via an IP-PAD that converts voice packet data to PCM signals. Call control signals are transmitted to the MP over the LAN, while voice packets are transmitted via the IP-PAD. The number of IP-PADs depends on the traffic volume of connections between the IP Enabled Dterm and legacy stations and trunks.

UNIVERGE NEAX 2000 IPS System Configuration

The *UNIVERGE* NEAX 2000 IPS consists of single or multiple Port Interface Modules (PIM) depending on the system configuration, and there are two types of PIM's; "Physical" PIM and "Virtual" PIM. The Physical PIM is "hardware" PIM which is used to accommodate an MP, FPs, IP PADs, legacy LT cards, AP cards, and power supply units. One Physical PIM provides up to 64 LT ports and up to 8 Physical PIM's can be accommodated in a Stand Alone system. A Virtual PIM is a "software" PIM and provides up to 64 ports per PIM for use by system programming as DtermIP telephones, Wireless PS stations or Peer to Peer (PTP) CCIS trunks. The system consists of up to 16 PIM's, by the combination of Physical PIM's and Virtual PIM's, thus providing 1020 ports. When the use of Virtual PIM's exceeds 8 then the number Physical PIM's is reduced by one for each additional Virtual PIM required.

The illustration below shows examples of 1020-port configuration by the combination of TDM LT ports, Dterm IP telephones, Wireless PS stations and Peer to Peer (PTP) CCIS trunks.



Module and Installation Hardware

The *UNIVERGE* NEAX 2000 IPS is comprised of up to 8 Port Interface Modules (PIM's). For installing the PIM's in various installation methods, additional installation hardware is required.

Modules

1.) Port Interface Module (PIM)

A PIM provides 13 card slots for common control, Line/Trunk (LT), and Application Processor (AP) cards. It also houses an AC/DC Power Supply, DC/DC Power Supply (for -48V), and batteries for protection from short-term (about 30 min.) power interruption.

Four champ connectors for Line/Trunk (LTC 0 to 3) are located at the lower front side of the PIM.

A PIM provides a maximum of 12 card slots for Line/Trunk (LT) and Application Processor (AP) cards. At maximum configuration, the system is comprised of 8 PIM's.

There are two types of PIM (PIMMJ and PIMMK) depending on the system type as follows.

Type of PIM	Single MP System	Dual MP System
PIM MJ	Used for PIM 0-7	Used for PIM 1-7
PIM MK	Not used	Used for PIM 0

PIM MJ (PIM3)	PIM MJ (PIM7)
PIM MJ (PIM2)	PIM MJ (PIM6)
PIM MJ (PIM1)	PIM MJ (PIM5)
PIM MJ (PIM0)	PIM MJ (PIM4)

(Single MP System)

PIM MJ (PIM3)	PIM MJ (PIM7)
PIM MJ (PIM2)	PIM MJ (PIM6)
PIM MJ (PIM1)	PIM MJ (PIM5)
PIM MK (PIM0)	PIM MJ (PIM4)

(Dual MP System)

Unit Configuration

2.) Battery Module (BATTMH)

The BATTM is an optional module for installing optional long-term (about 3 hours) backup batteries. The BATTM is designed to accommodate batteries covering up to a 4-PIM system (2 BATTMs support maximum system configuration).

The BATTM is available for Floor Standing Installation or 19 inch Rack-mounting. (When the system is Wall-mounted, the BATTM cannot be installed with the PIM.)

Modules

Abbrev	Description	Remarks
PIMMJ	SN1729 PIMMJ	Single MP System: PIM 0 - PIM 7 Dual MP System: PIM 1 – PIM 7
PIMMK	SN1730 PIMMK	Single MP System: Not used Dual MP System: PIM 0
BATTM	SN1731 BATTMH	1 per STACK, Max.2 per system

Installation Hardware

Base/Top Assembly

The Base/Top Assembly includes a Base Unit and a Top Cover for the PIM. One Base/Top Assembly is required for each PIM stack. The Base Unit also serves as the AC power distribution panel for up to a four PIM configuration.

Hanger Assembly

The Hanger Assembly is used for Wall-mounting Installation. One set of Hanger Assembly is required for each PIM.

19 inch Bracket

The 19-inch Bracket is a set of hardware used for 19-inch Rack-mounting Installation. The 19-INCH RACK BRACKET (A) is installed on both sides of the PIM. One set of 19 inch Bracket (A) is required for each PIM. The 19-INCH RACK BRACKET (B) is installed at the BASE of stack. One 19-INCH BRACKET (B) is required for each stack.

If the system is 2 PIM or more configurations with 19-INCH BRACKET (B), one set of 19-INCH BRACKET (A) is also required for the topmost PIM.

Optional Brackets

The Mounting Bracket is used for Floor Standing Installation. Without Mounting Bracket, 1.1G shockproof is provided for 1 to 3-module stack and 0.5G shockproof is provided for 4 or more module stack. To enhance the shockproof capability to 1.1G, one set of Mounting Bracket is required for each 4 or more module stack and attached to the topmost PIM.

The I/F Bracket is used for Floor Standing Installation to join the neighboring topmost PIM in 6 PIM or more configurations. One set of I/F Bracket is required for multiple stacks.

The Base Tray Assembly is used for Floor Standing Installation for stationary equipment (UL complied). One set of Base Tray Assembly is required for each stack.

Installation Hardware

Abbrev	Description	Quantity
Top Cover	TOP COVER ASSEM	1/STACK (BASE ASSEM is local supply)
Base/Top ASSEM	SN1545 BASERE	1/STACK
Hanger Assem	HANGER ASSEM (UL)	1/PIM (Wall-mounting Installation)
19 inch Bracket	19 INCH RACK BRACKET (A)	1/PIM (19 inch Rack-mounting Installation)
	19 INCH RACK BRACKET (B)	1/STACK (19 inch Rack-mounting Installation)
Mounting Bracket	MOUNTING BRACKET	OPTION (1/STACK)
I/F Bracket	I/F BRACKET ASSEM	OPTION (1/SYSTEM)
Base Tray	BASE TRAY ASSEM	OPTION (1/STACK)

UNIVERGE NEAX 2000 IPS SYSTEM POWER SUPPLY

AC/DC Power Supply

The AC/DC Power Card is mounted in the left side of each PIM. The AC/DC Power card provides power to all circuit cards, which reside in the PIM. AC power requirements are as follows:

Input Voltage: 90 to 132 Vrms or 180 to 264 Vrms (selectable by switch) 50/60 Hz

AC/DC Power Card

Name Code	Qty	Remarks
PZ-PW121	1 per PIM	AC/DC Power Supply Input:100 V/200 V Output:+5 V,-27 V,+90V, CR

DC/DC Power Unit

The DC/DC Power Unit is mounted under the AC/DC Power Card and generates -48 V power for the circuit cards that need such power.

DC/DC (-48V) Power Card

Name Code	Qty	Remarks
PZ-PW122	1 per PIM	DC/DC Power Supply, Input: -27 V, Output: -48 V

Battery Backup

Internal Short-term option

For customers requiring battery backup, short-term and/or long-term options are available. Two 3.4AH batteries are required per PIM, and installed inside of each PIM. Backup time is approx. 30 minutes when PHS (Wireless PS) is not accommodated and approx. 10 minutes when PHS (Wireless PS) is accommodated in the system.

Internal Battery Option

Name Code	Qty	Remarks
PWR CA-A	1 per (2) PIM's	DC Power Cable (-27v input)
BATT CA INT	1 per PIM	Internal Battery Cable

External Long-term option

Two 24AH batteries are required per each 2 PIM's, and installed inside of Battery Module in a stack basis. Backup time is approx. 3 hours when PHS (Wireless PS) is not accommodated and approx. 2 hours when PHS (Wireless PS) is accommodated in the system. The batteries are varied depending on the requested backup time. The battery shall be locally provided.

External Battery Option

Name Code	Qty	Remarks
PWR CA-A	(No. of PIM) –1= up to 4PIM (No. of PIM) –2= 5 to 8PIM	DC Power Cable (-27v input)
BATT CA EXT	1/Stack	External Battery Cable

Cabling

The internal and external cabling for the *UNIVERGE* NEAX 2000 IPS is as follows:

Internal Cabling

BUS Cable
The BUS Cable provides a connection between PIM's for I/O Bus, PCM Bus, and Alarm Bus extension in a multiple-PIM configuration.

Bus Cable

Name Code	Qty	Remarks
48-TW-0.7 CONN CA	(No. of PIM)-1	0.7 m (2.3ft.)

Power Control Cable
The Power Control Cable provides a connection between the MP and the AC/DC Power Supply for controlling battery backup function and MJ/MN alarm indication.

Power Control Cable

Name Code	Qty	Remarks
PWR CNT CA-D	1/PIM	Power control cable (for PIMl-PIM7): BWB - PZ-PW121/122 (Included in PIM)
PWR CNT CA-E	1/PIM	Power control cable (for PIM0) BWB - PZ-PW121/122 (Included in Base and PIM for Dual System)

DC Power Cable
The DC Power Cable provides a connection between the AC/DC Power Supplies for extending the 27 VDC input in a multiple-PIM configuration, when battery backup option is required.

Power Cable (Internal Battery Option)

Name Code	Qty	Remarks
PWR CA-A	1/2 PIM	DC Power Cable (-27v input)
BATT CA INT	1/PIM	Internal Battery Cable

Power Cable (External Battery Option)

Name Code	Qty	Remarks
PWR CA-A	(No. of PIM) –1= up to 4PIM (No. of PIM) –2= 5 to 8PIM	DC Power Cable (-27v input)
BATT CA EXT	1/Stack	External Battery Cable

AC Cord
The AC cord provides a connection between the AC/DC Power Card and Base Unit.

AC Cord

Name Code	Qty	Remarks
AC CORD-B-U	1/PIM	AC power cable (Included in PIM)
AC CORD-D-U	1/BASE	AC power cable for BASE (Included in BASE)

External Cabling

IP Connection

MP with PZ-M606-A and IP PAD

Name Code	Qty	Remarks
IP TRK BUS CA	1/VCT	IP TRK BUS Cable 0.1m (0.33ft) (Included in SPN-16VCTA IP PAD)

IP Trunk (H.323, CCIS over IP w/IVS2)

Name Code	Qty	Remarks
IP TRK BUS CA	1/VCT	IP TRK BUS Cable 0.1m (0.33ft) (Included in SPN-4VCTI-A, or SPN-4VCTI-B.)

Maintenance Administration Terminal (MAT) (RS-232C)

Direct Connection

MAT Cable (Direct)

Name Code	Qty	Remarks
MAT CA-T	1/MAT	RS-232C cable, 2m (6.6ft.)
MAT CA-P	1/MAT	RS-232C cable, 4m (13.1ft.)

Remote connection

Remote connection is available via an internal modem in the MP or an external modem.

MAT Cable for external modem

Name Code	Qty	Remarks
RS NORM-4S CA-A	1/MAT	RS-232C cable, 4m (13.1ft.)

Built-in SMDR/MCI on MP

The MP card provides RS-232C interface ports for SMDR/MCI connection.

- No. 0/1 Port: Async. : 1200/2400/4800/9600/19200 bps

Connection to SMDR is made on a port basis by a front cable.

Cable for Built-in SMDR/MCI on MP

Name Code	Qty	Remarks
RS RVS-4S CA-C	1/SMDR	RS-232C cable, 4m (13.1ft.) MP-DTE
RS NORM-4S CA-A	1/SMDR	RS-232C cable, 4m (13.1ft.) MP-DCE (MODEM)

SMDR/PMS/Hotel Printer/MCI (with AP00B MRC-C)

The PN-AP00-B card provides 4 RS-232C interface ports for SMDR, PMS, CIS Printer, Hotel Printer, MCI and Centralized SMDR. Connection to SMDR, PMS or Printer is made on a port basis by a front cable.

Cable for AP00

Name Code	Qty	Remarks
RS RVS-4S CA-C	1/SMDR or PMS	RS-232C cable, 4m (13.1ft.) PN-AP00 - DTE
RS NORM-4S CA-A	1/SMDR or PMS	RS-232C cable, 4m (13.1ft.) PN-AP00 - DCE (MODEM)
RS PRT-15S CA-A	1/Printer	RS-232C cable, 15m (49.2ft.) PN-AP00 - Printer

External Alarm Display

The *UNIVERGE* NEAX 2000 IPS can provide an optional external alarm display unit (ALM DSPP) to indicate Power-ON/OFF, Major Alarm, and Minor Alarm status.

External Alarm Display Panel

Name Code	Qty	Remarks
ALM DSPP	1/System	External Alarm Display Panel

Digital Trunk Interface (DTI)

Primary Rate Interface (PRI)

CCIS Trunk Interface (CCT)

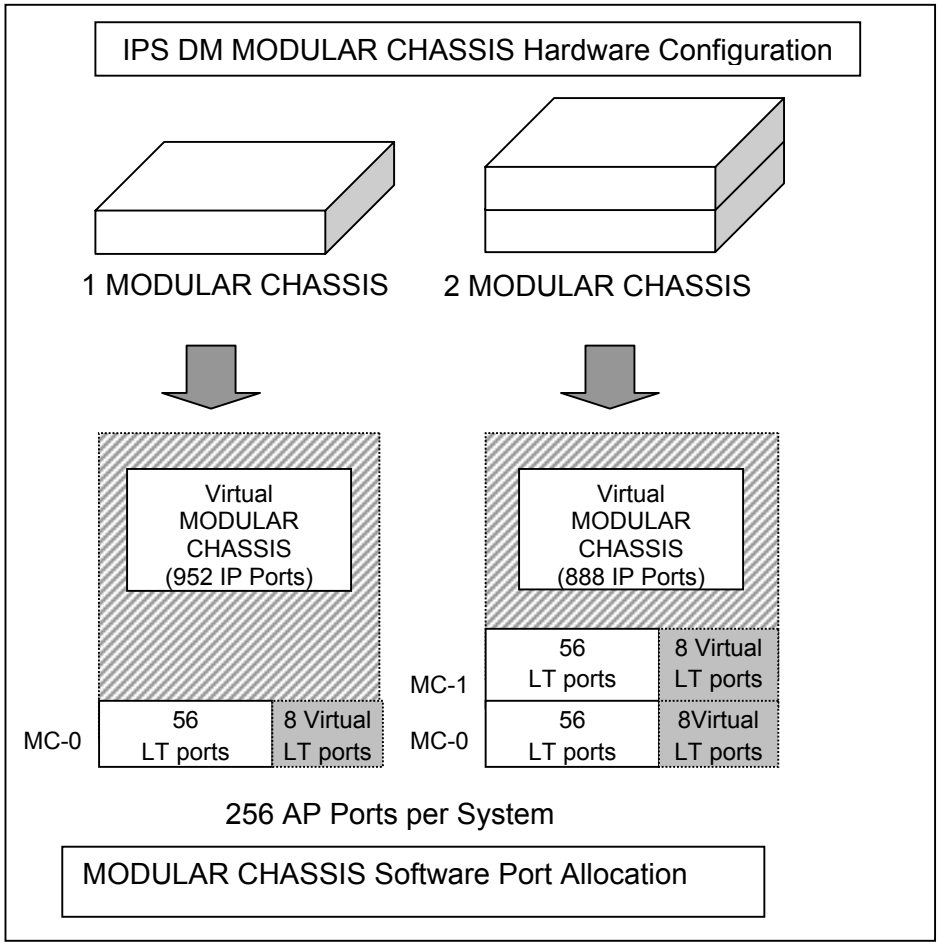
The DTI/PRI/CCT connection is available through twisted-pair cable via the MDF.

UNIVERGE NEAX IPS DM/IPS DML/IPS DMR System Configuration

IPS DM/IPS DML Modular Chassis (MC)

The *UNIVERGE* NEAX IPS DM consists of one to three MODULAR CHASSIS depending on the system configuration. The *UNIVERGE* NEAX IPS DML consists of one or two MODULAR CHASSIS depending on the system configuration. The MODULAR CHASSIS provides 56 LT ports in hardware slots and provides 64 ports in software port allocation (56 LT ports and 8 virtual ports). There are 2 types of MODULAR CHASSIS; "Physical MODULAR CHASSIS" and "Virtual MODULAR CHASSIS ". The Physical MODULAR CHASSIS is a “hardware MODULAR CHASSIS” and is used to accommodate an MP, IP PADs, legacy LT/AP cards, and power supply units. The Virtual MODULAR CHASSIS is a “software MODULAR CHASSIS” used to accommodate IP stations and Peer-to-Peer CCIS Trunks by system data programming. The port capacity of the Virtual MODULAR CHASSIS is varied depending on the number of Physical MODULAR CHASSIS.

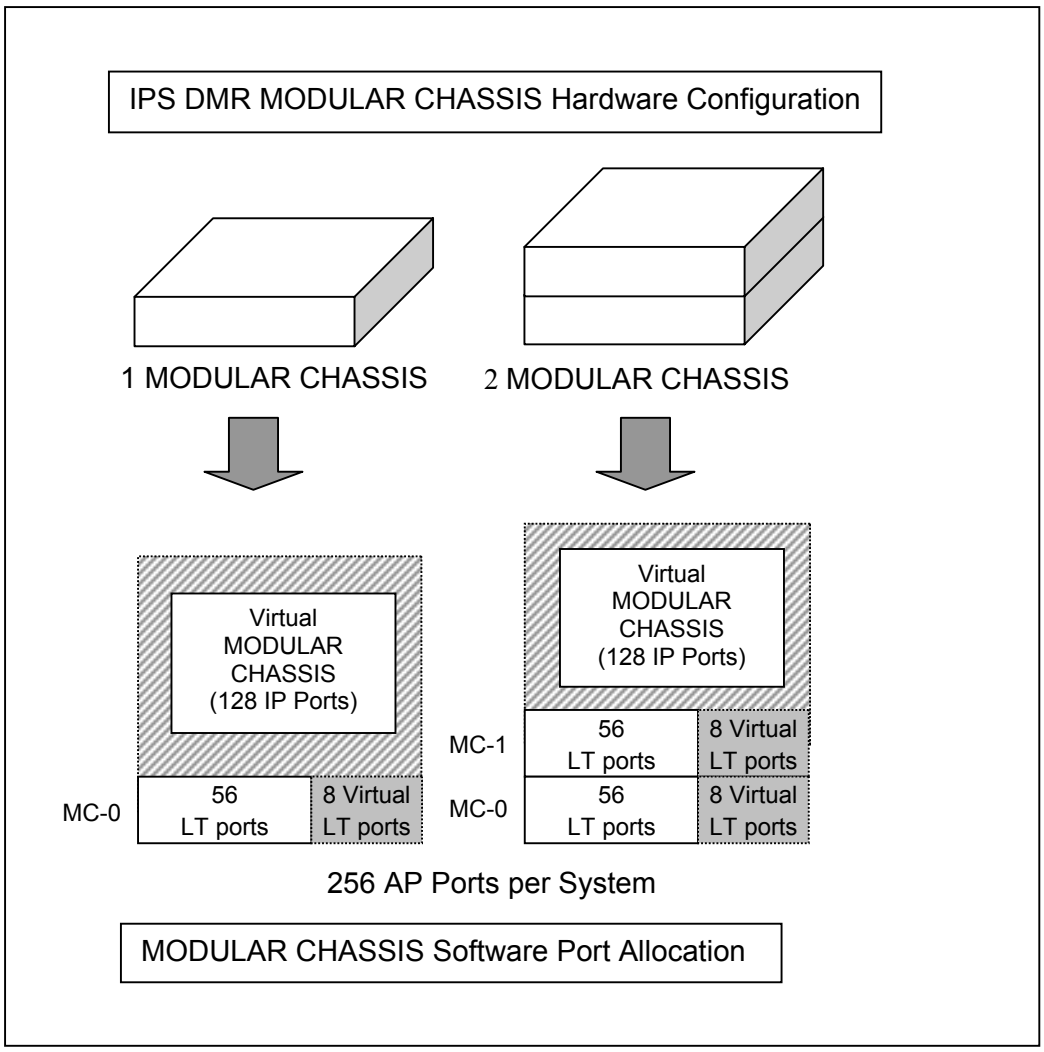
One MODULAR CHASSIS provides 8 card slots including one card slot for Main Processor (MP) and other 7 slots for Line Trunk (LT)/Application Processor (AP) cards; 56 LT ports and 8 virtual LT ports; AC, LTC, BUS cable connectors and power switch which are located at the rear side of MODULAR CHASSIS. The following illustration shows MODULAR CHASSIS hardware configurations, software port allocation, face layout and rear view of MODULAR CHASSIS for IPS DM/IPS DML.



IPS DMR Modular Chassis (MC)

There are 2 types of MODULAR CHASSIS; "Physical MODULAR CHASSIS" and "Virtual MODULAR CHASSIS ". The Physical MODULAR CHASSIS is a “hardware MODULAR CHASSIS” and is used to accommodate an MP, IP PADs, legacy LT/AP cards, and power supply units. The IPS DMR can consist of one or two MODULAR CHASSIS depending on the system configuration. The Physical MODULAR CHASSIS provides 56 LT ports in hardware slots and provides 64 ports in software port allocation (56LT ports and 8 virtual ports). The Virtual MODULAR CHASSIS is a “software MODULAR CHASSIS” with a port capacity of 64 ports. A maximum of two Virtual MODULAR CHASSIS can be assigned per remote site for a total of 128 ports used to accommodate IP stations by system data programming. The maximum number of Remote Sites is 30. This system allocates the maximum of 64 FP/AP cards per system, at multiple Remote Sites. The number of FP/AP cards accommodated at one Remote Site should be a maximum of eight including the MP with built-in FP. If more than eight FP/APs are assigned, the system does not operate normally.

The following illustration shows MODULAR CHASSIS hardware configurations, software port allocation, face layout and rear view of MODULAR CHASSIS for IPS DMR.



Modular Chassis (MC) Installation

The MODULAR CHASSIS can be installed on the desktop or into the 19-inch rack only.

Modular Chassis (MC) and Bracket

The following tables show the name and specification number of MODULAR CHASSIS and bracket. The bracket is selected depending on the installation method (Desktop or 19" Rack).

Modular Chassis

Description	Qty	Remarks
SN-1737 IPSMJ	N	Modular Chassis With Out RGU(Molded Cover)

Bracket

Description	Qty	Remarks
RACK MOUNT KIT(U)	N	1/Modular Chassis for 19” rack-mounting installation
JOINT BRACKET KIT(U)	N	1/Expansion Modular Chassis for desktop-setting installation

Cabling

BUS Cable / AC Cord

Description	Qty	Remarks
BUS-0.4 CA-X	1 Expansion MODULAR CHASSIS	BUS cable for 8 Slot Chassis
BUS-0.4 CA-CHG	1 Expansion MODULAR CHASSIS	BUS cable for connection between 6 Slot and 8 Slot Chassis
AC CORD-E-U	1 per MODULAR CHASSIS	AC Cord

Note: *When expanding an existing 6 slot Chassis only a two chassis configuration is supported and the 6 slot chassis must be PIM 0.*

PFT and Battery cable

Description	Qty	Remarks
BATT CA-P5	1 per MODULAR CHASSIS	PFT and Battery connection cable 5m (15ft). Cable for connecting Battery to 4PFT of AC/DC Power

System Power Supply and Cooling Fan

AC/DC Power Supply

The AC/DC Power Card is mounted in the rear side of each MODULAR CHASSIS. The AC/DC Power card provides power to all circuit cards, which are resided in the MODULAR CHASSIS.

AC power requirements are as follows:

Input Voltage: 85 to 264VAC, 50/60 Hz (Auto sensing 100V/200V selection)

AC/DC Power Card

Description	Qty	Remarks
PZ-PW131	1 per MODULAR CHASSIS	AC/DC Power Supply Output:+5 V, 5.5A, -27 V, 3A

Back-up Power

UPS or battery backup (with external battery) is available for backup power for AC power failure. In case of battery backup, sealed type battery should be used. The open type battery is not available because of AC/DC power specifications.

Required Battery Capacity

Battery Backup Time	Required Battery Capacity
10 minutes (Built-in battery)	12VDC, 0.8AH x 2
30 minutes (External battery)	12VDC, 2.2AH x 2
3 hours (External battery)	12VDC, 12AH x 2

Ring Generator Unit (RGU)

The RGU provides ringing supply to analog SLT (PN-4LCAA is no longer required); however, it cannot ring more than 6 analog SLT's per MODULAR CHASSIS simultaneously. Therefore, the simultaneous ringing has to be limited by office data setting.

Ring Generator Unit (RGU) Card

Description	Qty	Remarks
PZ-PW139	MODULAR CHASSIS	Ring Generator Unit Ring Voltage: 85/75Vac Ring Current: 0.05A

Cooling FAN

The cooling FAN is mounted at the right side of each MODULAR CHASSIS. The FAN should be replaced every 4 years in order to ensure the cooling system.

Cooling FAN

Description	Qty	Remarks
109P0624H 7D09 FAN	1 per MODULAR CHASSIS	Each Modular Chassis is equipped with a Cooling Fan this part is for replacement only

Software Architecture

The *UNIVERGE* NEAX 2000 IPS systems offer very attractive system starting points. Listed below are the System/Expansion Packages available with the *UNIVERGE* NEAX 2000 IPS family.

System Packages

***UNIVERGE* NEAX 2000 IPS System Packages**	
Description	**Remarks**
UNIV IPS IP 8 Seat System Package IPS UNIV PIMMJ (UA) ICS VS BASE-C (UA) SPN-CP24-C (CPU) PZ-M606-A SPN-8IPLA IP PAD-C 64 Port SYS SOFTWARE KEY KEEPER (FD) 8 SEAT LICENSE MAT*WorX* IPS	IP Starter Kit Package. Supports up to 8 IP Dterm's, 64 LT Ports, 5 T1's /E1's, 5 ISDN-PRI DCH's, 48 ISDN-BRI Trunks. NEC Customer Software License Agreement Required.
UNIV IPS 64 Port Basic System Package SPN-CP24-C (CPU) IPS UNIV PIMMJ (UA) ICS VS BASE-C (UA) 64 Port SYS SOFTWARE MATWorX IPS	Supports up to 64 LT Ports & 1 T1. NEC Customer Software License Agreement Required
UNIV IPS Back-Up CPU 8 IP Seat SYS PKG IPS UNIV PIMMK (UA) ICS VS BASE-C (UA) SPN-CP27A (CPU) SPN-CP27A (CPU) PZ-M606-A PZ-M606-A SPN-8IPLA IP PAD-C 64 Port SYS SOFTWARE KEY KEEPER (FD) 8 SEAT LICENSE KEY MAT*WorX* IPS	Package is equipped with 64 TDM ports and 8 IP. Provides matching CPU cards with the same CPU number. With matching CPU's, only one set of software is needed. NEC Customer Software License Agreement Required.
UNIV IPS Back-Up CPU 64 Port (TDM) SYS PKG (IPS) IPS UNIV PIMMK (UA) ICS VS BASE-C (UA) SPN-CP27A (CPU) SPN-CP27A (CPU) 64 Port SYS SOFTWARE MAT*WorX* IPS	Package is equipped with 64 TDM ports License for each CPU. Provides matching CPU cards with the same CPU number. With matching CPU's, only one set of software is needed. NEC Customer Software License Agreement Required.
IPS/IVS^2 UNIV Upgrade Kit SPN-CP24-C (CPU) IPS DM UNIV Front Cover 64 Port SYS SOFTWARE	Upgrades CPU to CP24 C and Adds UNIVERGE Front Cover NEC Customer Software License Agreement Required

***UNIVERGE* NEAX IPS DM/IPS DML System Packages**	
Description	**Remarks**
UNIV IPS DML IP SYS PKG IPS DM(E) UNIV PIMMJ AC CORD-E-U Rack Mount Kit SPN-CP31-C (CPU) PZ-M606-A SPN-8IPLA IP-PAD-C 64 Port SYS SOFTWARE Key Keeper (FD) 8 Seat Licenses 24 Port Patch Panel MAT*WorX* IPS	Provides for up to 48 Legacy ports and 128 IP stations per system. Comes with 8 IP Seat Licenses. Additional IP Seats Licenses are required beyond 8. NEC Customer Software License Agreement Required
UNIV IPS DM IP SYS PKG-E IPS DM(E) UNIV PIMMJ AC CORD-E-U Rack Mount Kit SPN-CP24-C (CPU) PZ-M606-A SPN-8IPLA IP-PAD-C 64 Port SYS SOFTWARE Key Keeper (FD) 8 Seat Licenses 24 Port Patch Panel MAT*WorX* IPS	Provides for up to 48 Legacy ports and 952 IP stations per system. Comes with 8 IP Seat Licenses. Additional IP Seat Licenses are required beyond 8. NEC Customer Software License Agreement Required
UNIV IPS DM/DML Expansion Module 1(E) IPS DM(E) UNIV PIMMJ AC CORD-E-U Rack Mount Kit Joint Bracket Kit BUS-0.4 CA-PA Key Keeper (FD) LT Port 64 24 Port Patch Panel	Adds 56 additional Legacy ports for a total of 112 (DM/DML) and supports up to 888 IP station (DM Only) per system. DML supports 128 IP stations max.
DM to UNIV DM Upgrade Kit SPN-CP24-C (CPU) IPS DM UNIV Front Mask 64 Port SYS SOFTWARE MAT*WorX* IPS	Upgrades to CP24C CPU and Adds UNIVERGE Front cover.

IP REMOTE PIM PACKAGES	
Description	**Remarks**
UNIV IPS DMR SYS PKG-E IPS DM(E) UNIV PIMMJ SPN-CP31-C (CPU) PZ-M606-A SPN-8IPLA IP PAD-C AC CORD-E-U Rack Mount Kit 24 Port Patch Panel	DMR Package for Remote PIM. Provides DM Chassis for Remote PIM over IP. The 8IPLA PAD card is provided. Remote PIM Site License required.
UNIV IP REMOTE PIM-B (IPS) IPS UNIV PIMMJ (UA) ICS VS BASE-C (UA) PN-CP24-C PZ-M606-A SPN-8IPLA IP PAD-C	IPS Package for Remote PIM. Provides IPS PIM that can be used as Remote PIM over IP. Remote PIM Site License required.
UNIV IPS DMR Expansion Module IPS DM(E) UNIV PIMMJ AC CORD-E-U Rack Mount Kit Joint Bracket Kit BUS-0.4 CA-PA 24 Port Patch Panel	Adds another DM PIMMF (Modular Chassis) to provide five card slots for 56 additional TDM ports.

Software Keys/Licenses

The *UNIVERGE* NEAX 2000 IPS systems offer both TDM Software Keys and Peer-to-Peer IP Seat Licenses, which are designed with modularity in mind. This approach allows customers a greater degree of cost-control for new installations and for upgrades to features, capacities and software series.

Description	Remarks
Generic Program	
64 Port Sys Software	Basic Business/Hotel/Motel Features for: 64 LT Ports, 5 T1's /E1's, 5 ISDN-PRI DCH's, 48 ISDN-BRI Trunks. NEC Customer Software License Agreement Required
Key Keeper (Parent Capacity Option)	
Key Keeper (FD)	Floppy Disk that holds selected Key files from below
Capacity Option (used w/Key Keeper)	
LT 64 Port	Expands LT Ports from 64 to 1020 Ports in increments of 64.
CCIS Link (1) CCIS Link (4) CCIS Link (8)	Adds support for one CCIS Link Adds support for four CCIS Link Adds support for eight CCIS Link
IPT Card (1) IPT Card (4) IPT Card (8)	Adds support for one IP trunk card Adds support for four IP trunk card Adds support for eight IP trunk card
Event Based CCIS (ECCIS) Key	Adds Event Based CCIS capability
Wireless Software Key	Adds Wireless with support of 128 ZT's and 256 PS's
Wireless Software Key 8 PS	Adds 8 Wireless ports each key above the initial 256
T1/E1 6 to 10	Expands T1/E1 capacity between 144 to 240 channels
ISDN DCH 5 to 8	Expands capacity between 5 DCH Cards and 8 DCH cards
IP Remote PIM 1 Site License	Adds IP Remote Capability and is required for each Remote Site
Dterm IP Software Seat Licenses	
8 IP Seat License Key	DtermIP terminals are controlled by the CPU and do not use digital line cards, Instead they require DtermIP seat licenses. The licenses are available in 8 seat increments and are cumulative. For example, if you have 8 existing DtermIP Seats and need a total of 16, add another 8-seat license for a total of 16 seats.
SP-30-4 Seat License	Each SP-30 4 Seat License can support up to four simultaneous sessions via the SP-30 Soft-Phone. The SP30 Soft-Phone also requires IP 8 Seat License. To support eight simultaneous soft-phone sessions would require eight Soft-Phone Licenses and 8 IP Seat Licenses.
SP-30-4 ACD Seat License	Each SP-30 4 ACD Seat License can support up to four simultaneous sessions via the SP-30 ACD Soft-Phone and can support either Business or ACD mode. The SP30/SP30 ACD Soft-Phone also requires IP Seat License. To support eight simultaneous soft-phone sessions would require eight Soft-Phone Licenses plus 8 IP Seat Licenses.

Chapter 3 System Highlights

Processors

The *UNIVERGE* NEAX 2000 IPS, IPS DM, IPS DML and IPS DMR are distributed multiprocessor systems. Their control system consists of a Main Processor (MP), Firmware Processors (FP), and Application Processors (AP). Both the FP and APs execute their predetermined functions under the control of the MP.

Main Processor (MP)

Name Code	Remarks
PN-CP24-C	Main Processor Card for *UNIVERGE* NEAX 2000 IPS and IPS DM. One card is required per system.
PN-CP27-A	Main Processor Card for *UNIVERGE* NEAX 2000 IPS Dual MP System. One card is required per system.
PN-CP31-C	Main Processor Card for *UNIVERGE* NEAX IPS DML One card is required per system.(Stand Alone system or CCIS end point)
	Main Processor Card for *UNIVERGE* NEAX IPS DMR. One card is required for each Remote Site.
PZ-M606	Ethernet Control Card: • Mounted on MP card to accommodate the Ethernet and transmit/receive a signal of TCP/IP protocol. • 10 BASE-T/100 BASE-TX twisted pair cable is connected directly to this card.

Major specifications and functionality of the *UNIVERGE* NEAX 2000 IPS MPs are shown below:

Item	PN-CP24-C PN-CP27-A	PN-CP31-C
Central Processing	ElanSC520	
System Memory	Flash ROM (16MB), SDRAM (64MB) **Note 1**	
Network Switching	1,024 · 1,024 Time Division Switch	
3-Way Conference	16 sets of 3-way conference circuitry	
DTMF Signal Sender	32 circuits (digit 0 to 9, *, and # are generated)	
Music-on-Hold	10 types are available **Note 2**	
Mini Jack	1 for External Music Source for Music on Hold **Note 2**	
Audible Tone Generator (DTG)	Available	
Phase Lock Oscillator (PLO)	2 ports (Source/Receiver)	
Built-in SMDR	Available	
Built-in MCI	Available	
Built-in FP0	Available	
BS00 Function	Available	
DTMF Receiver	4 circuits	
AP01 Function	Available	
Built-in DRS	Available	
MAT Interface	---	---
Direct Connection	1 port	1 port
Remote Connection w/Built-in MODEM	1 port	1 port **Note 3**
External Alarm Indication	MJ and MN	MJ only
DAT	2 circuits (120 seconds per circuit)	Not Available
DK00	2 circuits (relay drive x1, external key scan x1)	Not Available
Application Key Program	In EPROM	In Flash ROM
MP Program Download	Upgrade Program via FTP Server **Note 1**	

Note 1: *CP-27-A memory is 8MB Flash and 32MB SDRAM and MP Program Download is not supported.*
Note 2: *In case of Dterm IP, the synthesized melody is provided by the IP adapter on the phone*
Note 3: *Modem on the CP31 is for Stand Alone mode IPS DML only*

Firmware Processor (FP)

Firmware Processors (FP) are required when more than two PIM's are used. The FP provides supervision and status analysis of line/trunk ports, which reside in the PIM. The FP provides the bus interface for I/O Bus, PCM Bus, and Alarm Bus in a multiple-PIM configuration. The major specifications of the FP are shown below:

- Central Processor Unit: 16-bit (25 MHz)
- Memory: Program Area (384 kb), Work Area (384 kb)
- BS01 Function

Item	Remarks
PN-CP15	Firmware Processor Card for use with the UNIVERGE NEAX 2000 IPS.

Application Processor (AP)

Application Processors communicate directly to the MP, bypassing the FP. Features such as SMDR, Property Management System (PMS), Digital Trunks (T1), CCIS, etc. are Application Processors.

System Capacity for Application Processor Card:

- Maximum of 24 cards per system
- Maximum of 256 ports per system

There are several different AP cards. The table below provides a list of APs and their functions.

Application Processor (AP) Cards

AP CARD	Function
SPN-24DTAC (AP)	24 Channel Digital Trunk Interface with built-in CSU
SPN-30DTC (AP)	30 Channel Digital Trunk Interface with A/U Law conversion
SPN-24PRTA (AP)	24 Channel PRI Trunk with built-in D-Channel Handler (DCH)
SPN-24PRTA QSIG(AP)	24 Channel PRI Trunk with built-in QSIG D-Channel Handler (DCH)
SPN-24CCTA (AP)	24 Channel CCIS Trunk with built-in Common Channel Handler (CCH)
SPN-2BRTC (AP)	2 circuit BRI (Trunk)
SPN-4BRTA (AP)	4 circuit BRI (Trunk)
SPN-SC00 CCH (AP)	Common Channel Handler for CCIS Signaling
SPN-SC01 DCH (AP)	D-Channel Handler for ISDN Primary Rate Access
SPN-SC03 8ICH (AP)	D-Channel Handler for BRI
SPN-SC03 8CSH (AP)	CS (ZT) Handler (CSH)
SPN-SC01 QSIG (AP)	Q-SIG Protocol Handler
SPN-AP00B MRC-C (AP)	SMDR, PMS, H/M Printer & MCI Port
SPN-AP00B MRC-F (AP)	DND Group, PMS Executed History Print out, SMDR (2400 Format) & MCI
SPN-4RSTBA (AP)	4 circuit MF Receiver for T1 trunks with ANI
SPN-4RSTBA-911 (AP)	4 circuit MF Receiver for enhanced E911
SPN-4RSTC (AP)	4 circuit ID Number Receiver for Analog Caller ID (Class SM)
SPN-IPTB-A (AP)	IP Trunk for VoIP (P-MP)
SPN-IPTB-B (AP)	IP Trunk for VoIP (H.323)
SPN-CFTC (AP)	32-Party Conference
SPN-AP00B DBM (AP)	Data Base Memory, Wireless Roaming
SPN-SC01 DCH-Q (AP)	Q931a Protocol Handler for Wireless Roaming

Reliability and Availability

Mean Time Between Failures (MTBF)

The MTBF represents the minimum time frame in which a failure of any common equipment --such as processors, memory, switching, and etc. - may cause a partial system outage. The following table shows the MTBF in hours and in years for each card and terminal.

Reliability and Availability Chart

Description	Reliability				Availability
	Fit	MTBF years	MTBF Minute	MTTR Minute	
ICS VS PIMMJ (BWB)	1868	61.1	32136156	60	99.999813%
SPN-CP24A MP	11849	9.63	5064995	45	99.999112%
SPN-CP24B MP	10798	10.6	5575176	45	99.999193%
SPN-CP31A MP	10700	10.7	5627772	45	99.999200%
PN-CP27A MP	10798	10.6	5575176	0.3	99.999995%
SPN-CP15 FP	3060	37	19460520	5	99.999974%
PZ-M606-A	1269.2	89.94	47304842	10	99.999979%
PZ-PW121	13850	8.2	4312872	5	99.999884%
PZ-PW122	5780	19.8	10414008	10	99.999904%
PZ-PW131	13350	8.6	4523256	15	99.999668%
PN-8LCAA	1934	59	31031640	5	99.999984%
PN-8COTS	3312	34.5	19092348	5	99.999974%
PN-4DATC	5490	20.79	10934708	5	99.999954%
PZ-8PFTB	1439	79.3	41708628	5	99.999988%
PN-CFTB	941	121	63641160	5	99.999992%
PN-CFTC (AP)	6000	19.8	10414008	5	99.999952%
PN-IPTB-B (AP)	5757	19.83	10429787	5	99.999952%
SPN-4VCTI-B	1435.3	79.54	41834858	5	99.999988%
PN-4LCAA	4467	25.6	13464576	5	99.999963%

Reliability and Availability Chart (Cont.)

Description	Reliability				Availability
	Fit	MTBF Years	MTBF Minute	MTTR Minute	
PN-2ILCA	5849	19.5	10256220	5	99.999951%
PN-8DLCP	1848	61.77	32488549	5	99.999985%
PN-2DLCN	856	133.36	70142026	5	99.999993%
PN-M10	1750	65.23	34308371	5	99.999985%
PN-DK00	1274	89.6	47126016	5	99.999989%
PN-4LLCB	1792	63.7	33503652	5	99.999985%
PN-AP00B MRC-C (AP)	5196	22	11571120	5	99.999957%
PN-SC03-B 8ICH(AP)	6816	16.75	8809830	5	99.999943%
PN-16VCTAA IP PAD-A/B	2674	42.69	22453232	5	99.999978%
PN-20DTB	2965	38.5	20249460	5	99.999975%
SPN-32IPLAA IP PAD-C/D	11112	10.3	5417388	5	99.999908%
SPN-SC00 CCH-D(AP)	8816	12.95	6811182	5	99.999927%
PN-30DTCC-A(AP)	3504	32.58	17135777	5	99.999971%
SPN-24PRTA (AP)	4625	24.7	12991212	5	99.999962%
PN-24CCTA	4625	24.7	12991212	5	99.999962%
PN-8RSTG	133	858.31	4.51E+08	5	99.999999%
PN-4BRTA-F(AP)	6566.4	17.39	9146444	5	99.999945%
SPN-24PRTA-QSIG (AP)	4082	28	14726880	5	99.999966%
SN716 DESK CON-B	7790.4	14.3	7521228	5	99.999934%
PZ-M537	346	329.9	17409276	10	99.999943%
PZ-M542	1464	78	41024880	10	99.999976%
PZ-4PFTA	751	152	79945920	10	99.999987%
IPS DM PIMMF	13680	8.3	4365468	30	99.999313%

Reliability and Availability Chart (Cont.)

Description	Reliability				Availability
	Fit	MTBF years	MTBF Minute	MTTR Minute	
DTR-1-1 TEL	1359	84	44180640	5	99.999989%
DTR-1HM-1 TEL	2718	42	22090320	5	99.999977%
DTR-2DT-1 TEL	9927	11.5	6048540	5	99.999917%
DTR-8-1 TEL	8154	14	7363440	5	99.999932%
DTR-8D-1 TEL	10378	11	5785560	5	99.999914%
DTR-16D-1 TEL	10976	10.4	5469984	5	99.999909%
DTR-32D-1 TEL	12614	9.05	4759938	5	99.999895%
ITR-8D-2 TEL	14269	8	4207680	5	99.999881%
ITR-16D-2 TEL	14269	8	4207680	5	99.999881%
DCR-60-1P Console	9593	11.9	6258924	5	99.999920%
AD(A)-R Unit	2857	39.95	21012102	10	99.999952%
AP(R)-R Unit	6668	17.12	9004435	10	99.999889%
AP(A)-R Unit	6406	17.82	9372607	10	99.999893%
CT(A)-RP Unit	8154	14	7363440	10	99.999864%
IP-R Unit	12144	9.4	4944024	10	99.999798%
DCU-60-1	9593	11.9	6258924	5	99.999920%
DTP-8-1 TEL	8272	13.8	7258248	5	99.999931%
DTP-8D-1 TEL	10570	10.8	5680368	5	99.999912%
DTP-16D-1 TEL	11083	10.3	5417388	5	99.999908%
DTP-32D-1 TEL	12826	8.9	4681044	5	99.999893%
DTP-1-2	3110	36.7	19302732	5	99.999974%
DTP-1HM-2	4494	25.4	13359384	5	99.999963%

Mean Time To Repair (MTTR)

The Mean Time To Repair has been studied with the following conditions:

1. A mid to high-level technical person will attend to the site for the detection of the fault and replacement of the faulty circuit board.
2. All critical circuit boards for replacement are stocked at each Fault Dispatch Center (FDC).
3. All non-critical circuit boards for FDCs are stocked at NEC. The calculation result of the MTTR is 1.0 hour.

Reliability Calculations

The following information is to show you how to calculate the reliability for the system you are configuring. Remember that these calculations are used to determine a partial system outage.

Reliability Calculation

$$1FIT = \frac{1 Defect}{1{,}000{,}000 Hours \bullet Parts} \qquad MTBF = \frac{1 FIT}{Total\ FITS} = \frac{10^9}{Total\ FITS}$$

System Calculation

To calculate the MTBF of a single system, obtain the sum of the FITs from each component.
For example: Assume you have a complete system that consists of two components:

Item A FIT=31,700, Item B FIT=80,000
Total Fit = 31,700 + 80,000 = 111,700

$$MTBF = \frac{10^9}{111{,}700} = 8952.55 \text{ Hours}$$

$$\frac{8{,}952.22\ Hours}{8{,}760 Hours / Year} = 1.02\ Years$$

Availability Calculations

$$Availability = \frac{MTBF}{MTBF+MTTR} = .9XXXXX$$

System Traffic

In regards to the traffic capacity of electronic PABX, two kinds of descriptions are usually employed. One is Busy Hour Call Attempts (BHCA), which relates to the ability of the Central Processor Unit (CPU). Another is the traffic handling capacity per line that relates to the handling capacity of each station. This technical information herein offers the traffic data for both cases.

Traffic Load

Traffic load is expressed as the quantity of traffic during the busy hour, the busiest one-hour period of the day for traffic. Traffic can be measured in minutes, hours, hundreds of call seconds (CCS), or Erlangs.

CCS– one CCS is equal to 100 seconds of telephone time, or 1/36 Erlang.

Erlang– one Erlang is equal to one hour of telephone time or 36 CCS.

CCS units are convenient when traffic is measured in small increments. For telephone traffic, a typical business generates about 6 CCS per telephone during the busy hour (1/6 Erlang).

Busy Hour

The hour during the day when traffic is highest is called the busy hour. Since this is the period when users are most likely to encounter blockage, the traffic load during the busy hour is the load you need to plan for. Traffic tables show the amount of traffic that can be handled during the busy hour. Within the busy hour, peaks will occur when blockage will be higher than the design objectives. It's not economical, however, to design a network to handle absolute peaks. Most networks are designed to cause some traffic to be blocked or overflow to other services.

One way to establish the load during the busy hour is to schedule traffic studies for a week when you know the load is high. Or, you can use tables developed by traffic engineers to estimate the busy hour load.

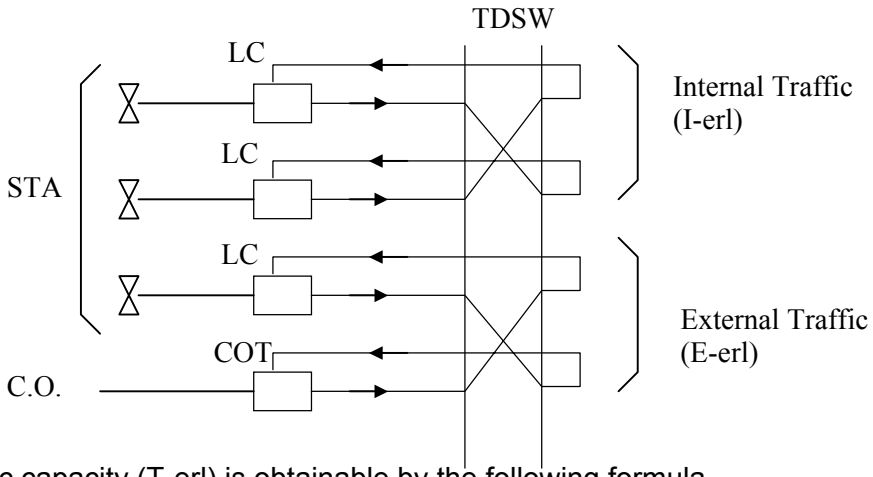
BHCA (Busy Hour Call Attempts)

BHCA of the *UNIVERGE* NEAX 2000 IPS is shown below.

Number of PIM's	1PIM	2PIM	3PIM	4PIM	5PIM	6PIM	7PIM	8PIM
Business or Hotel/Motel without ACD/OAI	Max. 2500 BHCA (No FP)		Max. 5000 BHCA (One FP)		Max. 7500 BHCA (Two FPs)		Max. 8000 BHCA (Three FPs)	
Business or Hotel/Motel with ACD/OAI	Max. 2000 BHCA (No FP)		Max. 4000 BHCA (One FP)		Max. 5000 BHCA (Two FPs)		Max. 5500 BHCA (Three FPs)	

Traffic Capacity In Erlang (For TDM Circuits)

Calculation Procedure



Total system traffic capacity (T-erl) is obtainable by the following formula.

T (erl) = I (erl) + E (erl)

In the *UNIVERGE* NEAX 2000 IPS, external traffic capacity is obtained from the Central Office Trunk (COT) quantity by employing the “Erlang B Table”. There is no limitation for the internal traffic capacity. The time slot is assigned for individual station and the connection between stations can be made without limitation through TDSW (Time Division Switch). Consequently, when describing the traffic capacity in Erlang, it is meaningless unless specifying the ratio of the internal and external traffic. From the above idea, total system traffic capacity (T-erl) is actually obtained by the following formula:

T (erl) = Internal Traffic (I-erl) + External Traffic (E-erl)
= E (erl) , (Ratio of external traffic)
When assuming the external and internal traffic is even.
T (erl) = E (erl) + 0.5

For example:
System configuration:

- 384 Lines
- 64 C. O. Trunks
- External traffic: Internal traffic: = 7:4

In this system, external traffic capacity is obtained from “Erlang B” table. That is, 64 C. O. Trunks at grade of service of 0.01 can carry 50.6 erl. Total traffic capacity of this system is obtained from above 50.6 erl and ratio of external traffic as shown below:

T (erl) = E (erl) ÷ (Ratio of external traffic)

$$= 50.6 \div \frac{7}{7 \div 3}$$

= 72.3 erl

Traffic capacity per station line
= 72.3erl ÷ 384 lines
= 0.19 erl/L

Traffic Capacity per Line (For TDM Circuits)

Traffic capacity per line (erl/L) can be calculated in accordance with the calculation procedure in Section 3.4. The traffic capacity per line at various line/truck configurations as the calculation result is shown in the following table. Though the following table does not cover all *UNIVERGE* NEAX 2000 IPS line/trunk configurations, it covers the typical TDM line/trunk configurations likely to be required in the actual system. Grade of service is considered as 0.01. Use the Erlang B Table to find the required number of DTMF register ports for a specific *UNIVERGE* NEAX 2000 IPS.

Unit = erl per line

Number of Trunks

Use Erlang B table to determine the number of trunks at a particular grade of service after using the following formulas:

$$\text{Incoming:} \quad \frac{\text{incoming ccs traffic}}{36 \text{ ccs}} = \text{Erlang B}$$

$$\text{Outgoing:} \quad \frac{\text{outgoing ccs traffic}}{36 \text{ ccs}} = \text{Erlang B}$$

Note: *Traffic loads are measured in hundred call seconds, or CCS. Since there are 3,600 seconds in an hour, a line that's tied up for one hour measures 36 CCS traffic.*

Erlang B Table

Traffic Distribution		Internal Call: 0.3	:0.4	:0.5	:0.6	:0.7
		External Call: 0.7	:0.6	:0.5	:0.4	:0.3
Line	Trunk					
16L	4 COT	0.078	0.091	0.109	0.136	0.181
	8 COT	0.279	0.326	0.391	0.489	0.652
	12 COT	0.525	0.612	0.735	0.918	1.224
32L	4 COT	0.039	0.045	0.054	0.068	0.091
	8 COT	0.140	0.163	0.196	0.244	0.326
	12 COT	0.262	0.306	0.367	0.459	0.612
	16 COT	0.396	0.462	0.555	0.693	0.925
48L	4 COT	0.026	0.030	0.036	0.045	0.060
	8 COT	0.093	0.109	0.130	0.163	0.217
	12 COT	0.175	0.204	0.245	0.306	0.406
	16 COT	0.264	0.308	0.370	0.462	0.613
96L	12 COT	0.087	0.102	0.122	0.153	0.204
	16 COT	0.132	0.154	0.185	0.231	0.308
	20 COT	0.179	0.209	0.251	0.313	0.418
	24 COT	0.228	0.266	0.319	0.398	0.531
144L	12 COT	0.058	0.068	0.082	0.102	0.136
	16 COT	0.088	0.103	0.123	0.154	0.205
	20 COT	0.119	0.139	0.167	0.209	0.278
	24 COT	0.152	0.177	0.212	0.266	0.354
192L	20 COT	0.090	0.104	0.125	0.157	0.209
	24 COT	0.144	0.133	0.159	0.199	0.266
	28 COT	0.139	0.162	0.194	0.243	0.324
	32 COT	0.164	0.191	0.230	0.287	0.383
240L	20 COT	0.072	0.084	0.100	0.125	0.167
	24 COT	0.091	0.106	0.127	0.159	0.212
	28 COT	0.111	0.129	0.155	0.194	0.259
	32 COT	0.131	0.153	0.184	0.230	0.306
288L	20 COT	0.060	0.070	0.084	0.104	0.139
	24 COT	0.076	0.089	0.106	0.133	0.177
	28 COT	0.093	0.108	0.129	0.162	0.216
	32 COT	0.109	0.128	0.153	0.191	0.255
336L	28 COT	0.079	0.092	0.111	0.139	0.185
	32 COT	0.094	0.109	0.131	0.164	0.219
	36 COT	0.108	0.127	0.152	0.190	0.253
	40 COT	0.123	0.144	0.173	0.216	0.288
	44 COT	0.138	0.161	0.194	0.242	0.323
384L	28 COT	0.069	0.081	0.097	0.121	0.162
	32 COT	0.082	0.096	0.115	0.144	0.191
	40 COT	0.108	0.126	0.151	0.189	0.252
	44 COT	0.121	0.141	0.169	0.212	0.282
	48 COT	0.134	0.157	0.188	0.235	0.313
	64 COT	0.188	0.220	0.264	0.329	0.439

PEG Count

The PEG counter is used for maintenance purposes to verify various data in the system. PEG counters can be set by system programming to start and stop by month, day, hour and minute. A system reset will clear all PEG count data. The following data can be measured by PEG count.

Trunk PEG Count

- Number of outgoing trunk seizure by Trunk Route
- Number of tandem connections established
- Number of times a busy station was encountered
- Number of all types of calls to Attendant Console
- Number of connections giving Dial Tone
- Number of station-to-station connections established
- Number of failures caused by all senders being busy
- Number of failures caused by all registers being busy
- Number of failures caused by all ringing trunks being busy
- Number of failures caused by all IP-PAD channels being
- Number of forced release of communication between station and Trunk/Tandem connection
- Number of call forwarding caused by the calling number is not informed from network
- Number of rejection of the incoming call the calling is not informed from network
- Number of incoming call seizure-Trunk Route
- Number of times all trunks found to be busy trunk route
- Number of incoming calls terminated to busy tone-Trunk Route
- Number of unanswered incoming calls-Trunk Route
- Number of register connection on trunk call-Trunk Route
- Number of conference calls (Three/Four way Calling)
- Number of failures cased by all conference trunks (For three way Calling) being busy
- Number of transferred incoming calls to Attendant Console or predetermined station, by Call Forwarding-Don't Answer

UCD PEG Count

- Number of answered calls on UCD station
- Number of incoming calls to UCD Group
- Number of call waiting calls for predetermined time in queuing mode on UCD Group
- Number of abandoned calls to UCD Group
- Number of incoming calls to all busy of UCD Group
- Number of incoming calls to UCD Group that were answered
- Number of times queuing

Wireless PEG Count

- Number of Call Origination
- Number of Call Termination
- Number of Location Registration
- Number of Handovers
- Number of Out of Cell

IP Network PEG Count

- Number of times that traffic exceeded the limit bandwidth
- Number of times that traffic exceeded the warning bandwidth
- Maximum bandwidth used
- Bandwidth used now

Number of Voice Channels per IP Trunk

The maximum voice channels per IP Trunk card depends on the payload size as follows (payload size can be assigned in system programming):

CCIS (Point-to-Multipoint)

Payload Size	G.729a	G.711	G.723.1
20 ms	8 Channel	8 Channel	------
30 ms	16 Channel	16 Channel	16 Channel
40 ms	16 Channel	16 Channel	------

VoIP (H.323)

Payload Size	G.729a	G.711	G.723.1
20 ms	6 Channel	5 Channel	---
30 ms	8 Channel	7 Channel	8 Channel
40 ms	12 Channel	10 Channel	---

Payload size for Virtual IPT

Payload Size	Max. Voice Channels Per IPT		
	G.729a	G.711	G.723.1
10 ms.	4	4	–
20 ms.	8	8	–
30 ms.	16	16	16
40 ms.	16	16	–

IP PAD Calculation

IP PAD Calculation								
	Number of extensions	**8**	**16**	**32**	**48**	**72**	**96**	**100**
Legacy:	Number of PAD Ports	5	6	9	12	16	20	20
0%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	5	6	9	12	15	19	19
10%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	4	6	9	11	15	18	19
20%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	4	6	8	11	14	17	18
30%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	4	6	8	10	13	16	16
40%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	4	5	7	9	12	15	15
50%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	4	5	7	8	11	13	13
60%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	3	4	6	7	8	10	10
75%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	3	4	5	6	8	9	9
80%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	2	3	4	5	5	6	6
88%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	2	2	3	3	4	5	5
95%	Number of IP PAD Cards	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	3	3	4	5	6	7	7
99.0%	Number of IP PAD Cards	1	1	1	1	1	1	1

IP PAD Calculation

	Number of extensions	120	200	208	264	300	320	376	432
Legacy:	Number of PAD Ports	23	34	35	43	48	50	57	64
0%	Number of IP PAD Cards	1	2	2	2	2	2	2	2
Legacy:	Number of PAD Ports	22	33	34	41	46	48	55	62
10%	Number of IP PAD Cards	1	2	2	2	2	2	2	2
Legacy:	Number of PAD Ports	21	31	32	39	44	48	52	59
20%	Number of IP PAD Cards	1	1	1	2	2	2	2	2
Legacy:	Number of PAD Ports	20	30	30	37	41	43	49	55
30%	Number of IP PAD Cards	1	1	1	2	2	2	2	2
Legacy:	Number of PAD Ports	19	27	28	34	37	40	45	51
40%	Number of IP PAD Cards	1	1	1	2	2	2	2	2
Legacy:	Number of PAD Ports	17	25	25	30	34	35	40	45
50%	Number of IP PAD Cards	1	1	1	1	2	2	2	2
Legacy:	Number of PAD Ports	15	22	22	27	29	31	35	39
60%	Number of IP PAD Cards	1	1	1	1	1	1	2	2
Legacy:	Number of PAD Ports	11	16	17	20	22	23	26	28
75%	Number of IP PAD Cards	1	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	10	14	14	17	19	20	22	24
80%	Number of IP PAD Cards	1	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	7	9	10	11	12	12	14	15
88%	Number of IP PAD Cards	1	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	5	6	7	7	8	8	9	10
95%	Number of IP PAD Cards	1	1	1	1	1	1	1	1
Legacy:	Number of PAD Ports	8	11	11	13	14	14	16	18
99.0%	Number of IP PAD Cards	1	1	1	1	1	1	1	1

System Specifications

AC Power Requirements

Description	Specifications
AC Input Voltage	90 to 132Vac or 180 to 264Vac; 47 to 64Hz
AC Input Current	3.5A(at 100V), 2.0A(at 200V)

AC Power Consumption / Thermal Output (Maximum)

Description	AC Power Consumption (KVA)		Thermal Output (BTU)	
	100V	200V	100V	200V
1-PIM	0.35	0.40	1,195	1,365
2-PIM	0.70	0.80	2,389	2,730
3-PIM	1.05	1.20	3,584	4,096
4-PIM	1.40	1.60	4,778	5,461
5-PIM	1.75	2.00	5,973	6,826
6-PIM	2.10	2.40	7,167	8,191
7-PIM	2.45	2.80	8,362	9,556
8-PIM	2.80	3.20	9,556	10,922

Battery Requirements

Description	Specifications
Max. Battery Capacity	260AH per 4 PIM (65AH (12V) x 8)
DC Input Voltage for Battery	-24V
Built-in Battery Requirements	3.4AH (12V) x 2 (approx. 30min. backup)
Physical Size of Built-in Battery (one 12V battery)	133(W) x 60(H) x 67(D) mm

Operating Environment

Description	Specifications
Ambient Temperature	0° to 40° Celsius
Relative Humidity	Max. 90% (non-condensing)

Electrical Characteristics (Central Office Trunk)

Description	Specifications
Insulation Resistance	15 mega-ohms or more at 100Vdc
DC Resistance	On-hook conditions: 30 mega-ohms Off-hook conditions: 1,700 ohms
Impedance	On-hook conditions: 20 kilo-ohms (300 to 3,400Hz) 8 kilo-ohms (at 24Hz) Looped conditions: 600 ohms
Leak Current	0 mA at on-hook conditions

Transmission Characteristics (For TDM Circuits)

Description	Specifications
PCM Coding System	A-law/U-law
Insertion Loss	0.15 dB at 1KHz
Return Loss	20 dB or more (300 to 3,400Hz) against 600 ohms
Longitudinal Balance	59 dB or more (300 to 3,400Hz)
Attenuation/Frequency Distortion	-0.2 dB to +0.7 dB (300 to 3,400Hz)
Group Delay Distortion	0 to 0.3msec. (500 to 2,800 Hz)
Total Distortion	25 dB (Input signal:-45 dBm0)/40 dB(input signal:0 dBm0)
Idle Channel Noise	-67 dBmop or less (psophometric noise) -50 dBm0 or less (single frequency noise)
Impulsive Noise	0 counts at -35 dBm
Cross Talk Attenuation	90 dB or more
Inter-modulation Products	-40 dB or more
Spurious In-Band Signals	-49 dBm0 or less
Signal Attenuation	Attenuation rate: 12 dB per octave or more at 3.4 kHz above Attenuation level: -40 dBm or less at 3.4 kHz and above -70 dBm or less at 50 kHz and above

Line Conditions

Description		Specifications
Loop Resistance (including Telephone Set)		
Analog Standard Line		Max. 600 ohms
Analog Long Line		Max. 2,500 ohms (DP 10pps), Max. 1,700 ohms (DP 20pps) Max. 1,200 ohms (DTMF)
Loop Resistance (including Opposite End Resistance)		
Central Office Trunk		Max. 1,700 ohms
Tie Line Trunk (Loop Dial)		Max. 2,500 ohms
Tie Line Trunk (E&M)		Max. 900 ohms (only E-wire condition)
Cable Length Note 1		
SN716 Desk Console		
	8DLC/4DLC/2DLC Card	Max. 350 meters (Max. 300 meters for 8DLC card)
	4DLC/2DLC Card with AC Adapter	Max. 1,200 meters
Dterm Series i/E		
	8DLC/4DLC Card	Max. 200 meters (Max. 300 meters for Dterm 8 and Dterm 8D)
	2DLC Card	Max. 850 meters
	4DLC/2DLC Card with AC Adapter	Max. 1,200 meters
DSS/BLF Console		
	4DLC/2DLC Card with AC Adapter	Max. 1,200 meters
Zone Transceiver		
	ZTII-U (for U-Interface)	Max. 1,210 meters (2-wire), Max. 2,100 meters (4-wire) @-48V Max. 970 meters (2-wire), Max. 1,700 meters (4-wire) @-45V
	ZTII-U with AC Adapter	Max. 3,950 meters

Note 1: *Cable length is based on cable with 0.5mm diameter and without lightning arresters*

Rotary Dial Pulse and DTMF Signaling

(1) Rotary Dial Signal

Description	Specifications	
	Receiving	Sending
Dial Speed	9 to 22 pps	10 pps +/- 0.8pps 20 pps +/- 0.8 pps
Break Ratio	55 to 77 %	67 +/- 3% or 62 +/- 3%
Inter-Digit Pause	Min. 256 msec.	300 to 1,000 msec.(10 pps)
Switch-Hook Flash Detection	384 to 2,300 msec.	Not applicable

(2) DTMF Signal

Description	Specifications				
	Receiving	Sending			
Signal Code	High Frequency Group		1,209Hz	1,336Hz	1,477Hz
	Low Frequency Group	697Hz	1	2	3
		770Hz	4	5	6
		852Hz	7	8	9
		941Hz	*	0	#
Frequency Deviation	+/- 1.8 %	+/- 0.8 %			
Signal Duration	Min. 40 msec.	64 or 128 milli-sec.			
Inter-Digit Pause	Min. 40 msec.	32 to 240 msec.			
Signal Level	-46 to -5 dBm	-10 dBm (low group) - 8 dBm (high group)			
Unwanted Frequency Components	Not Applicable	40 dB below the power of signal frequency			

Multi-frequency Compelled (MFC) – R2 SIGNAL

(1) MFC Frequency Value

Frequencies	Forward Signals(Hz)	Backward Signals(Hz)
F0	1,380	1,140
F1	1,500	1,020
F2	1,620	900
F3	1,740	780
F4	1,860	660
F5	1,980	540

(2) MFC Combinations

Combination Number	Frequencies
1	F0 + F1
2	F0 + F2
3	F1 + F2
4	F0 + F3
5	F1 + F3
6	F2 + F3
7	F0 + F4
8	F1 + F4
9	F2 + F4
10	F3 + F4
11	F0 + F5
12	F1 + F5
13	F2 + F5
14	F3 + F5
15	F4 + F5

(3) Sender/Receiver Specifications

Description	Specifications
Sender	
▪ Sender Transmitted Level	-8 dbm to -11.5 dBm
▪ Frequency Variation	+/- 2 Hz
Receiver	
▪ Sensitivity Range	-35 dBm to 0 dBm
▪ Frequency Variation	+/- 12 dBm

Audible Tones and Ringing Signal

Audible Tones

Tone	Frequency	Interruption
Dial Tone (DT)	350 Hz mixed with 440 Hz	Continuous
Special Dial Tone (SDT)	350 Hz mixed with 440 Hz	0.125 sec. ON, 0.125 sec. OFF
Busy Tone (BT)	480 Hz mixed with 620 Hz	0.5 sec. ON, 0.5 sec. OFF
Reorder Tone (ROT)	480 Hz mixed with 620 Hz	2.5 sec. ON, 0.25 sec. OFF
Howler Tone (HWT)	2,400 Hz interrupted by 16 Hz	Continuous
Service Set Tone (SST)	350 Hz mixed with 440 Hz	Continuous
Ring Back Tone (RBT)	440 Hz mixed with 480 Hz	1 sec. ON, 3 sec. OFF
Hold Tone (HDT)	480 Hz mixed with 620 Hz	0.25 sec. ON, 0.25 sec. OFF 0.25 sec. ON, 1.25 sec. OFF
Second Dial Tone	440 Hz mixed with 480 Hz	0.25 sec. ON, 0.25 sec. OFF 0.25 sec. ON, 1.25 sec. OFF
Call Waiting Ring back Tone	440 Hz mixed with 480 Hz	1 sec. ON, 1 sec. OFF
Message Waiting Indication Tone	350 Hz mixed with 440 Hz	0.125 sec. ON, 0.125 sec. OFF

Ringing Signal

Description	Specifications
Frequency	20 or 25Hz
Voltage	75 or 90Vrms

Note: *The IPS has the capability to detect the above type of signal from Central Office and to transmit the above type of signal to PBX stations.*

Dimension and Weight

***UNIVERGE* NEAX 2000 IPS**

Description	Specifications	
	Dimensions (W x D x H; mm)	Weight (kg)
Main Equipment		
▪ PIM (Fully card-mounted)	Approx. 430 x 223 x 353	Approx. 11.5
▪ BASE	Approx. 430 x 205.2 x 61.6	Approx. 3.0
▪ BASE TRAY (for UL)	Approx. 435 x 224.6 x 66.2	Approx. 1.7

***UNIVERGE* NEAX IPS DM/IPS DML/IPS DMR**

Description	Specifications	
	Dimensions (W x D x H; mm)	Weight (kg)
Main Equipment		
NEAX IPS DM NEAX IPS DML NEAX IPS DMR	430(W) x 365(D) x 88(H) mm	Approximately 7Kg / MODULAR CHASSIS (when all slots are occupied)

UNIVERGE NEAX IPS DM/IPS DML/IPS DMR System Specifications

System Specifications

Item	Specifications
System Capacity	LT ports: Max. 56 ports / MODULAR CHASSIS, (Max. 64 ports including 8 virtual LT ports/MODULAR CHASSIS) Max. 112 ports / system AP ports: Max. 256 ports / system IP ports: Max. 956 ports (IPS DM), Max. 128 ports(IPS DML/DMR) /system Card slots: 8 slots / MODULAR CHASSIS (including 1 slot for MP/FP card)
Circuit Card Mounted in MODULAR CHASSIS	All LT/AP cards of the *UNIVERGE* NEAX 2000 IPS can be used for the IPS DM/IPS DML/IPS DMR with the exception of 4LLCB, 4ODTA, and 4CSIA cards.
Power	AC100V – 240V (automatically adjusted)
Installation Method	Desk top-setting, 19” rack-mounting
Conditions	Temperature: 5°C – 40°C (when the system is operating) Humidity: 20% - 80% (when the system is operating)
Cooling	Cooling by FAN
Safety Standard	Complied with UL60950, CSA22.2 No. 950, EN60950, AS3260
EMC	Complied with VCCI Class A, FCC Part 15 Class A, EN55022 Class A, AS/NZS 3548 Class A

Open Industry Standards

Open Industry Standards supported by the *UNIVERGE* NEAX 2000 IPS	
G.711	IP Precedence
G.726(Fax only)	Differentiated Services
G.729a	RTP
H.323 V2	RTCP
Q.931	Ipv6 (In-skin router card required(RTA))
802.1d	H.225
802.1p	H.245
802.1q	TCP/IP
802.3	UDP/IP
SNMP	DHCP
Fax-Group 3	DNS

IP Specifications

Item	Specifications		Remarks
Voice Encoding	G.729a G.723.1 (5.3 k/6.3 k) G.711		8 kbps CS-ACELP MP-MLQ/ACELP 64 kbps PCM
IP-PAD	8 or 32 channels per card Automatically seized per call		
FAX Communication Feature	FAX Relay Method (T.30)		32 IPLA-PAD card is required. G3 FAX (up to 14.4 kbps) Super G3 Reciprocal: Not supported
	FAX Relay Method (Pass-thru FAX Method)		8 IPLA-PAD-B card w/R9 software required (32 IPLA not supported)
DTMF Signal	H.245		H.323 IPT/IP-PAD/DtermIP
Inter-office/Intra-office Signaling	H.245		DtermIP to DtermIP connection DtermIP to IP-PAD connection
	PROTIMS over IP		DtermIP to IPS connection
	CCIS over IP		Point to Multipoint connection
	H.323		H.323 IPT card and IP-PAD card are required
Jitter Control	Dynamic Jitter Buffer		
QoS (Quality of Service)	• TOS, IP Precedence • DiffServ		
LAN Interface	10BASE-T/100BASE-TX		Auto Negotiation is available. 100BASE-TX is recommended.
Echo Canceller (IP-PAD)	G.168		
Payload Size	DtermIP/CCIS Virtual IPT	10 ms-40 ms (G.723.1: 30 ms unit)	Max. voice channels per card 10 ms: 12 ch 20 ms: 20 ch 30 ms: 30 ch 40 ms: 32 ch
	H.323 IPT	20 ms.-40 ms. (10 ms. increments) (G.723.1: 30 ms. fixed)	Maximum voice channels per card G.729a / G.711 / G.723.1 20 ms.: 6ch / 5ch / - 30 ms.: 8ch / 7ch / 8ch 40 ms.: 12ch / 10ch / -
PAD Control		0 dB to +16 dB (+2 dB unit) 0 dB to –16 dB (–2 dB unit)	Setting is available per Location No.
		0 dB to –16 dB	For connection via the IPT card

LAN interface transmission modes

Device type	Auto negotiation	100 Mbps fixed mode		10 Mbps fixed mode	
		Full Duplex	Half Duplex	Full Duplex	Half Duplex
MP (M606)	X (Default)	X	—	—	—
32IPLA/32IPLA-A	X (Default)	X	X	—	X
8IPLA	X (Default)	X	—	—	—

X = available — = not available

System Compliance

HIPAA Summary

The security standards set forth within HIPAA require encryption; authentication and audit trail measures to safeguard patient medical information during electronic data interchange (EDI) transactions between healthcare providers and third party reimbursement entities. These measures impact how patient medical information is exchanged, as well as accessed within a facility's data network. Claims transmissions using Internet, intranets, extranets or private data networks are subject to HIPPA security standards as previously outlined.

HIPAA Compliance

Patient medical data is not accessed, stored or otherwise exchanged over a healthcare facility's private branch exchange (PBX). However, an NEC manufactured private branch exchange does provide the flexibility to create detailed call accounting records that can be utilized to audit or otherwise track utilization of a facility's NEC telecommunications system and/or PBX. A NEC PBX can also be configured to restrict access to designated telephone extensions. Furthermore, an NEC PBX can also be configured to utilize password access to individual telephones, voice mailboxes, etc. for purposes of end user authentication.

So, under the guise of the security standards within HIPAA, **NEC Private Branch Exchange equipment is HIPAA compliant** and does provide the necessary security measures for use within any business setting that is subject to administrative rules outlined with the Health Insurance Portability and Accountability Act.

Department of Defense Compliance

The NEC NEAX 2000 Internet Protocol Switch (IPS) with software release D1.8.20, hereinafter referred to as the system under test (SUT), meets all of its critical interoperability requirements and is certified for joint use within the Defense Switched Network (DSN) as a Private Branch Exchange (PBX) 2. *However, since PBX2s do not support the Military Unique Feature Requirements detailed in reference (c), connectivity to the DSN is not authorized until a waiver is granted by the CJCS.* PBXs are Military Department (MILDEP) controlled elements of the Defense Switched Network (DSN). PBX2 switches have no military unique features (MUFs) and can only serve Department of Defense (DOD), non-DOD, non-governmental, and foreign government users having no missions or communications requirement to ever originate or receive Command and Control (C2) communications. C2 users and Special C2 users are not authorized to be served by a PBX2. This interoperability test status is based on evaluation of Chairman of the Joint Chiefs of Staff (CJCS) validated Generic Switching Center Requirements (GSCR) for PBX2s and the overall system interoperability performance. The SUT also offers a Voice over Internet Protocol capability; however this capability is not covered under this certification.

This statement was obtained from a Department of Defense document posted on there web site and full copy of the compliance document can be downloaded from the following link.

http://jitc.fhu.disa.mil/tssi/apl.html

Chapter 4 Equipment List

Module/Installation Hardware

This table shows the names and functions of the modules.

Equipment Name	Function Name	Remarks
SN1729 PIMMJ	PIM	Port Interface Module (PIM) Maximum 64 physical ports per PIM. Houses two batteries for protection from short power interruption (for 30 minutes). At maximum configuration, the system consists of eight PIM's, and provides a total of 512 physical ports (64 ports × 8).
SN1730 PIMMK	PIM	Port Interface Module (PIM) for Backup CPU System Maximum 64 physical ports per PIM. Houses two batteries for protection from short power interruption (for 30 minutes). One PIM is required per Backup CPU System.
COVER PARTS ASSEM-A	COVER PARTS ASSEM	Cover Parts Assembly One cover parts assembly is required for each PIM.
SN1545 BASE	BASE/TOP ASSEM	Base/Top Cover Assembly One base and top cover assembly is required for each stack.
SN1731 BATTMH	BATTM	Battery Module for housing PIM or CS (ZT) backup batteries Houses two pairs of batteries for protection from long power interruption (for 3 hours).

This table shows the name and functions of installation hardware.

Equipment Name	Remarks
HANGER ASSEM (B)	Wall Hanger Assembly One HANGER ASSEM is required per PIM for Wall Mounting Installation.
MOUNTING BRACKET	Safety Mounting Bracket Used as an overhead hanger for Floor Standing Installation. Wire, chain or eyebolts are to be locally provided, to secure the bracket. To be installed on the top PIM in four or more modules of stack. It provides 1.1G shockproof construction.
19” RACK BRACKET (A)	19-inch Rack Mounting Bracket Type A One bracket is required for one PIM configuration. One bracket is required for the top PIM of multiple module configurations.
19” RACK BRACKET (B)	19-inch Rack Mounting Bracket Type B One bracket is required for the bottom module of multiple module configurations.
I/F BRACKET ASSEM	Inter Frame Bracket Assembly Used to join the frames in two-stack configuration, for Floor Standing Installation.
BASE TRAY ASSEM	Base Tray Assembly One BASE TRAY is required per one frame for Floor Standing Installation of Stationary Equipment.

Circuit Cards

The circuit cards used for *UNIVERGE* NEAX 2000 IPS systems are divided into the following types. According to these card types, the mounting locations of card and port allocation of the Time Division Switch are varied.

Common Control Cards
- Main Processor (MP)
- Firmware Processor (FP)
- Ethernet
- Power

Line/Trunk (LT) Cards
- IP PAD, Line Circuit (LC), Central Office Trunk (COT), Tie Line Trunk (LDT/ODT), etc.

Application Processor (AP) Cards
- SMDR/PMS/MCI/Hotel Printer Interface (AP00)
- T1/E1 Digital Trunk Interface (DTI)

Common Control Cards

The following table shows the names and functions of each control card.

Common Control Cards

Equipment Name	Function Name	Remarks
PN-CP24	MP	MP Card for Single MP System Provides LAN control function, system-based Device Registration Server (DRS), built-in FP, built-in OAI, built-in SMDR, Virtual IPT, Virtual CSH, 33 MHz PCI BUS, Memory (SDRAM 64 MB, Flash ROM 16 MB), TDSW (1024CH × 1024CH), 16-line CFT, PB sender, Clock, 2-line PLO (receiver mode/source mode), two RS-232C ports, 2-line DAT (Recording duration: Maximum 128 seconds), DK, 4-line PB receiver, Modem for remote maintenance (33.6 Kbps), Music-on-Hold tone, BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing, cable delay timing, monitors I/O Bus and PCM BUS. Supports MP Program Download. One card is required per system.
PN-CP27	MP	Main Processor Card for Backup CPU system Provides LAN control function, system-based Device Registration Server (DRS), Virtual FP, built-in OAI, built-in SMDR, Virtual IPT, Virtual CSH, 33 MHz PCI BUS, Memory (SDRAM 32 MB, Flash ROM 9 MB), TDSW (1024CH × 1024CH), 16-line CFT, PB sender, Clock, 2-line PLO (receiver mode/source mode), two RS-232C ports, 2-line DAT (Recording duration: Maximum 128 seconds), DK, 4-line PB receiver, Modem for remote maintenance (33.6 Kbps), Music-on-Hold tone, BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing, cable delay timing, monitors I/O Bus and PCM BUS. Two cards are required per Backup CPU system. One card for active MP [MP0] and another card for stand by MP [MP1].

Common Control Cards (Cont.)

PN-CP31	MP	Main Processor Card for Remote PIM (DML/DMR) Provides LAN control function, System-based Device Registration Server (DRS), built-in FP, 33 MHz PCI BUS, Memory (SDRAM 64 MB, Flash ROM 16 MB), TDSW (1024CH × 1024CH), 16-line CFT, PB sender, Clock, 2-line PLO (receiver mode/source mode), one RS-232C port, 4-line PB receiver, internal Music-on-Hold tone, BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing, cable delay timing, monitors I/O Bus and PCM BUS. Supports MP Program Download. One card is required per system.
PN-CP15	FP	Firmware Processor Card for the IPS. Provides Line/Trunk interface, Memory (RAM 768 KB), and inter-module BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing, cable delay timing, monitors I/O Bus and PCM BUS. When the system consists of three PIM's or more, one each of this card is mounted respectively in PIM2, PIM4, and PIM6. 3/4 PIM System: 1 per System 5/6 PIM System: 2 per System 7/8 PIM System: 3 per System
PZ-M606	ETHER	Ethernet Card, 1 per MP For Peer to Peer Connection and MAT/OAI Mounted on MP card to accommodate the Ethernet and transmit/receive a signal of TCP/IP protocol. Provides Auto Negotiation function (Always set to ON). 10 BASE-T/100 BASE-TX cable is connected directly to this card.
PN-PW00	EXTPWR	Power Supply Card for DESKCON Provides –48 V DC power. Maximum 3 cards per PIM. One card per two Desk cons Occupies two physical slots width per card.
PZ-PW121	AC/DC PWR	Main Power Supply Card Input: AC120 V/240 V (50 Hz/60 Hz) Output: –27 V (4.4 A), +5 V (7.2 A), CR (38 mA), +90 V (80 mA) One card is pre-installed per PIM.
PZ-PW122	DC/DC PWR	Power Supply Card for CS (ZT) and PN-4LLCB for Long Line Input: –24 V DC Output: –48 V DC (1.7 A) One card per PIM. Maximum 16 ZT's per card. Requires Ferrite Core for Long Line function

Application Processor Cards

The following table shows a summary of the Application Processor cards for *UNIVERGE* NEAX 2000 IPS systems.

Maximum 24 cards per system
Maximum 256 ports per system

Application Processor Card Name and Function

Function	Name	Port	Remarks
Digital Trunk Interface (DTI/PRT/BRT)	SPN-24DTAC-B (AP)	24	T1 Digital Trunk Interface (1.5 Mbps) Card: Accommodates 24-channel PCM digital lines. (Available up to 256 AP ports) T1 w/CSU
	SPN-30DTC-C (AP)	30	E1 Digital Trunk Interface (2 Mbps) Card: Accommodates 30-channel PCM digital lines. (Available up to 128 AP ports)
	SPN-24PRTA-D (AP)	25	ISDN Primary Rate (23B+D) Interface Card w/CSU: Provides a built-in D-Channel Handler (DCH) and built-in CSU
	SPN-24CCTA-A (AP)	25	CCIS (1.5 Mbps) Trunk Card: Provides a built-in Common Channel Handler (CCH) of CCIS.
	SPN-2BRTC (AP)	4	2-line Basic Rate (2B+D) Interface Trunk Card: Accommodates two two-channel PCM digital lines.
	SPN-4BRTA-F(AP)	8	4-line Basic Rate (2B+D) Interface Trunk Card: Accommodates four two-channel PCM digital lines. Point to Point Connection Only

Application Processor Cards (Cont.)

Function	Name	Port	Remarks
Common Channel Handler for NO.7 CCIS	SPN-SC00 CCH-D (AP)	1	Common Channel Handler for CCIS (1/Trunk Route). Transmits/receives signals on the common signaling channel of CCIS.
D-Channel Handler for ISDN PRI	SPN-SC01 DCH-C (AP)	1	D-Channel Handler Card: Transmits/receives signals on the D-Channel of ISDN Primary Rate (23B+D) interface.
D-Channel Handler for ISDN Station	SPN-SC03B 8ICH (AP)	4	ISDN-channel Handler Card: Provides the D-Channel signaling interface and controls a maximum of four ILC cards (Layer 2 and 3).
ZT Handler	SPN-SC03B 8CSH-C(AP)	4	CS (ZT) Handler Card: Provides the D-Channel signaling interface and controls a maximum of two CSI cards, eight CS (ZT) s. Supports Dukane Nurse Call system.
SMDR/MCI/PMS/CIS /Hotel Printer Interface	SPN-AP00B MRC-C(AP)	2	Application Package for SMDR (2400 basic and 1400 only/ PMS/ Hotel Printer/ CCIS Centralized SMDR Default data supported.
	SPN-AP00B MRC-F(AP)	2	Application Package for SMDR (2400 basic and extended only)/MCI/DND Automatic Set/Reset at Appointed Time and PMS Executed Task Print out.
	PZ-M537 **Note 1**	0	Memory Expansion Card for AP00 Card: • The system capacity is expanded when PZ-M537 is mounted on PN-AP00-B (AP00) card.
MFC Receiver/Sender	SPN-4RSTBA-B (AP)	4	4-line MF Receiver, MFC Receiver/Sender Card: • Used for MF/MFC-R2 Signaling on DID/DOD trunks. • A maximum of four cards can be provided per system, including the PN-4RSTC card.
	SPN-4RSTBA-911 (AP)	4	4 circuit MF Receiver for Enhanced 911 Requires CAMA Trunks
	SPN-4RSTC-A (AP)	4	4 circuit ID Number Received for Caller ID

Application Processor Cards (Cont.)

Function	Name	Port	Remarks
Wireless Roaming	SPN-AP00B DBM-C (AP)	2	Database Memory for Wireless Roaming and OAI Applications using Free Location Facility (FLF). 1 Card / system
	SPN-SC01 DCH-Q (AP)	1	DCH Card for Wireless Roaming 1 Card / Tie Line Route Installed with DTI Card
QSIG	SPN-SC01 QSIG (AP)	1	QSIG Protocol Handler
	SPN-24PRTA-QSIG (AP)	25	ISDN Primary Rate (23B+D) Interface Card w/CSU and built-in QSIG D-Channel Handler (DCH). Also provides Name Display.
32-Party Conference	SPN-CFTC (AP)	32	32-party Conference Trunk 8 cards per system max. **Note 2**

Note1: *The Expansion Memory (PZ-M537) is the on-board sub-memory card for the AP00-B. It provides additional memory capacity as shown below.*
Note2: *Each CFTC reduces system trunk capacity by 32*

	No EXPMEM		With EXPMEM	
Type of PN-AP00-B	**Local Office of Centralized Billing-CCIS /Stand-alone is Provided.**	**Center Office of Centralized Billing-CCIS is Provided. Local Office of**	**Centralized Billing-CCIS /Stand-alone is Provided.**	**Center Office of Centralized Billing-CCIS is Provided.**
PN-AP00-B with MRC-C program	1600	800	27000: Call Record for CIS is not provided. 26000: Call Record for CIS is provided.	
PN-AP00-B with MRC-F program	2620	1965	23580	22925

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Line/Trunk (LT) Cards

The following table shows a summary of the Line/Trunk (LT) cards for *UNIVERGE* NEAX 2000 IPS. The LT cards may be installed in slot 00 to 11 of PIM 0-7, with the following conditions:

Line/Trunk (LT) Cards

Function	Name	Port	Remarks
Analog Station	PN-8LCAA	8	8 Line Analog Station Card Message Waiting Control: 8 circuits Momentary Open: 8 circuits Loop Resistance: Max. 600ohms including telephone set
	PN-8LCAD	8	8 Line Analog Station Card Caller-ID for Analog Station Loop Resistance: Max. 600 ohms Message Waiting Lamp control Momentary Open for each circuit
	PN-8LCS	8	8 Line Analog Station Card Momentary Open: 8 circuits, Message Waiting notification via Stutter Dial Tone only. Loop Resistance: Max. 600 ohms
	PN-4LCD-A	4	4 Line Analog Station Card with MW Lamp control, Momentary Open: 4 Circuits On-Board power (80V) is included. Loop Resistance: Max. 600ohms including telephone set
	PN-AUCA	2	2 Line Analog Universal Trunk 2 Line Long Line Circuit with PFT Loop resistance: max. 2500ohms (LLC) Two reverse circuits is included On Board power (-48V) is included
	PN-4LLCB	4	4 Long Line Analog Station Card Works in either mode: Long Line Station (Requires Ferrite Core) Caller ID (Requires PN-4RSTF-A) Loop Resistance: Max. 2500ohms including telephone set PZ-PW122 is required. Can **not** be used in IPS DM or IPS DMR

Line/Trunk (LT) Cards (Cont.)

Function	Name	Port	Remarks
ISDN Station	SPN-2ILCA	8	2 Line ISDN Line Circuit Card • Provides a physical interface to ISDN Terminals. • Occupies eight time slots per one card.
Digital Station	PN-8DLCL	8	8-line Digital Line Circuit Card for Dterm Series i/E/III, Dterm Elite, IPK, DSS Console, DESKCON [–27 V version, two-wire type]
	PN-8DLCP	8	8-line Digital Line Circuit Card for Dterm Series i/E/III, DSS Console, DESKCON [–27 V version, 2-wire type]
	PN-4DLCM	4	4-line Digital Line Circuit Card for Dterm Series i/E, Dterm Elite, DSS Console, DESKCON [–27 V version, 2-wire type] Provides Line Test function.
	PN-4DLCQ	4	4-line Digital Line Circuit Card for Dterm Series i/E, DSS Console, DESKCON [–27 V version, 2-wire type] Provides Line Test function.
	PN-4DLCT	4	4-line Digital Long Line Circuit Card for Dterm Series i/E, DSS Console [–48 V version, two-wire type] PZ-PW122 required
	PN-2DLCN	2	2-line Digital Long Line Circuit Card for Dterm Series i/E, Dterm Elite, DSS Console [–48 V version, two-wire type]: • Provides Line Test function. • Equipped with –48 V DC-DC on-board power supply. Max. 850 m (2788.7 ft.)

Line/Trunk (LT) Cards (Cont.)

Function	Name	Port	Remarks
IP PAD (IP Station Peer-to-Peer)	SPN-32IPLAA IP PAD-E	32	32-channel IP-PAD card: • Provides Packet Assembly/Disassembly to accommodate Legacy Line/Trunk interface. • Used with a maximum of two 16VCT cards. • Two cards can be accommodated per built-in FP/FP card, a maximum of eight per system. • 100 BASE-TX twisted pair cable is connected directly to this card. • Supports T.30 or Pass-Through Fax (G.711/G.726). 16VCT is not required for G.711, 16VCT is required for G.726. (R9 enhancement). • Auto-Negotiation or 100Mbps Full Duplex is selectable.
	SPN-8IPLA IP PAD-C	8	8-channel IP-PAD Card Provides LAN Interface, Packet Assembly/ Disassembly to accommodate Legacy Line/Trunk interface. And provides voice compression DSP control functions such as voice compression control, DTMF relay. • Voice compression protocols: G.711 (64 kbps), G.723.1 (5.3 Kbps/6.3 Kbps), G.729a (8 Kbps) • FAX protocol: Pass-through (G.711, G.726) • Auto-Negotiation or 100Mbps Full Duplex is selectable. When mounting 24DSP (PZ-24IPLA) card, this card can provides up to 32-channel of IP-PAD (When using G.723.1, provides up to 24-channel of IP-PAD). Two cards can be accommodated per built-in FP/FP card, maximum 8 per system. 10BASE-T/100BASE-TX twisted pair cable is connected directly to this card.
	PZ-24IPLA	24	24-channel IP PAD and VCT daughter board. Mounts on SPN-8IPLA to provide up to 32 G.711, G.729a and 16 G.723.1 PAD channels. (T.30/T.38 is not supported)
	SPN-16VCTAA IP PAD-B	0	16-channel CODEC Card for IP-PAD: • Used with IP-PAD card. • Two cards can be accommodated per IP-PAD card, maximum 16 per system. • Voice compression protocols: • G.723.1, G.729a, G.711, • FAX (14.4 Kbps), DTMF signals.

Line/Trunk (LT) Cards (Cont.)

Function	Name	Port	Remarks
Analog C.O. Trunk	PN-8COTS	8	8-line Central Office Trunk Card (Loop Start/ Ground Start Trunk): Provides loop detection, sending/detecting ground on Tip/Ring wire.
	PN-8COTQ	8	8-line Central Office Trunk Card (Loop Start Trunk): Provides loop detection, receiving/sending the Caller ID (CLASS SM) signal.
	PN-4COTB	4	4-line Central Office Trunk Card (Loop Start/Ground Start Trunk): Provides loop detection, sending/detecting ground on Tip/Ring wire.
	PN-4COTG	4	4-line Central Office Trunk Card (Loop Start Trunk): Provides loop detection, receiving/sending the Caller ID (CLASS SM) signal.
Analog Tie Line Trunk	PN-2ODTA	2	2-line Out Band Dialing Trunk Card: Used as either a 2-wire E&M trunk or a 4-wire E&M trunk. Equipped with –48 V DC-DC on-board power supply. Both No. 0 and No. 1 circuits must be set to the same purpose (2-wire or 4-wire) in one card.
	PN-4ODTA	4	4-circuit analog 2W/4W Tie line card. The PZ-PW122 (-48V Power Supply) is required in case of E&M Type-I signaling.
Analog DID Trunk	PN-4DIDA	4	4-line Direct Inward Dialing Trunk Card: • Provides loop detection, sending reverse signal and PB to DP signal conversion.
	PN-AUCA	2	2L ANALOG UNIVERSAL TRUNK Used as 2-Line Long Line Circuit with PFT or 2-Line DID. Loop Resistance: MAX.2500ohms (LLC) Two Reverse Circuits Included On-Board Power (-48VDC) included
DTMF Receiver	PN-8RSTG	8	8-line PB Receiver Card: Used for a PB station line, DID or tie line.
Caller ID Sender	PN-4RSTF-A	4	4-line Sender Card for Caller ID Display on analog single line telephones: PN-4LLCB is required. Max. 4/sys.

Line/Trunk (LT) Cards (Cont.)

Function	Name	Port	Remarks
Conference Trunk	PN-CFTB	10	6-/10-Party Conference Trunk with Amp 6-party (1 CFTB card): 10 ports 10-party (2 CFTB cards): 20 ports
Digital Announcement Trunk	PN-2DATA	4	2-circuit Digital Announcement Trunk (Max. 60 sec. per channel)
	PN-4DATC	8	4-circuit Digital Announcement Trunk (Max. 120 sec. per channel)
Tone Trunk	PN-TNTA	4	2-line Tone/Music Source Interface Card: • Used for BGM or Music on Hold. • Provides two jacks for an external tone/music source.
Power Failure Transfer	PZ-8PFTB	0	8-line Power Failure Transfer Card: • Mounted in PFT slot of PIM. • One card per PIM.
	PN-AUCA	2	2 Line Analog Universal Trunk card that provides 2 Lines of Power Failure Transfer
IP Trunk	SPN-IPTB-A	4	IP Trunk card for VoIP (CCIS/ p-mp). This card is used together with a max of four PN-4VCT cards (16 channels) for connection with 2000/2400 non peer-to-peer connection. 1 card/PIM, 8 cards (Max. 127ch)/system
	SPN-4VCTI-A	4	4-channel CODEC Card for IP Trunk. Voice compression protocols: • G.723.1, G.729a, G.711, • FAX (14.4 Kbps), DTMF signals. • Used with PN-IPTB card.
	SPN-IPTB-B	4	VoIP H.323 IP Trunk card. Supports up to 12 channels. (G.729a. PAYLOAD 40ms). Also requires CCIS and IPT key per IPTB and NEC GK1000 software. Has one RJ45 connectors for 10/100 Ethernet/Fast Ethernet to router.
	SPN-4VCTI-B	4	4-channel voice translator CODEC card for H.323 includes BUS cable. Voice compression: G729A /G723/G711. Max 3 VCT's per SPN-IPTB-B

Line/Trunk (LT) Cards (Cont.)

Function	Name	Port	Remarks
Wireless Zone Transceiver interface	SPN-4CSIA	16	4-circuit CS interface for U-interface ZT. Two slot card, 16 LT ports. The PZ-PW122 is required.
Fiber Optical	PN-M10	0	Optical Fiber Interface Card: Provides optical fiber interface for T1/E1 Digital Trunk Interface (1.5 M/2 Mbps) or Remote PIM. Line length: 10 km (6.2 miles) or less. Line coding: CMI.
In-Skin Router	PN-RTA	0	In-Skin Router Card 10/100BASE-TX:1, 10BASE-T:1, RS2332C(D-sub 9pin) Maximum 8 cards per PIM
	PZ-M649	0	T1 Digital Trunk Interface (1.5Mbps) Card Mounted on PN-RTA Card Provides a built-in CSU.
	PZ-M623	0	Ether Control Card Mounted on PN-RTA Card Provides one additional 10BASE-T port
NEAXmail IM-16LX	PZ-VM07	4/8	4 or 8 Port Voice Mail card (NEAXmail IM-16LX) This card can provide up to 16 ports of digital line circuit interface. Occupies VM slot and up to one LT slot One card per PBX is available.
	PZ-VM08	8/16	4 Port Voice Mail card used to expand from 8 to 16 ports. Provides one connector for Occupies VM slot plus and one LT slots
External Relay Control	PN-DK00	0	External Relay Contact Control/Scanning (8-circuit/card) Works in either mode: Control / Scanning 8-circuit / 0-circuit 4-circuit / 4-circuit 0-circuit / 8-circuit

Chapter 5 Station Equipment

Terminal Line Up

A variety of terminal equipment may be connected to the *UNIVERGE* NEAX 2000 IPS systems.

Dterm Series i Terminals

Analog

- DTR-1-1 (Single-Line Analog)
- DTR-1HM-1 (Hospitality Single-Line Analog)
- DTH-1-1 (Low End Single Line Analog)

Dterm Series i (IP) Multi-line Digital Terminals

- ITR-4D-3 (4-Line display)
- ITR-8D-3 (8-Line display)
- ITR-16D-3 (16-Line display)
- ITR-16LD-3 (16-Line Desi-less display)
- ITR-32D-3 (32-line display)

Dterm Series i (TDM) Multi-line Digital Terminals

- DTR-2DT-1 (2-Line non display)
- DTR-4D-1 (4-Line non display)
- DTR-8-2 (8-Line non-display)
- DTR-8D-2 (8-Line with display)
- DTR-16D(BL)-2 (16-Line with Back lit LCD display)
- DTR-16D-2 (16-Line with display)
- DTR-16LD-2 (16-Line Desi-less display)
- DTR-32D-1 (32-Line with display)
- DCR-60-1 Console (Add-On Module/DSS/BLF)
- D16(LD)-R ADM (16 button Desi-less Adapter)

Dterm Series E Terminals

Multi-line (TDM) Digital Terminals

- DTP-2DT-1 (2-Line Digital Terminal)
- DTP-8-1 (8-Line Digital Terminal)
- DTP-8D-1 (8-Line Digital Terminal with Display)
- DTP-16D-1 (16-Line Digital Terminal with Display)
- DTP-32D-1 (32-Line Digital Terminal with Display)
- DTP-32DA-1 (32-Line Digital Terminal with Display for ACD)
- DCU-60-1 CONSOLE (Add-On Module/DSS/BLF)

Dterm Cordless Terminals

- DTR-4R-2 (Dterm Cordless II Terminal)
- DTH-4R-1 (Dterm Cordless Lite II Terminal)
- Plantronics CT Cordless

Inaset

- ITR-240G-1 (3" gray scale LCD and 16 programmable keys)
- ITR-320C-1 (5.1" color LCD and 32 programmable keys)
- ITR-320G-1 (5.1" gray scale LCD and 32 programmable keys)

Wireless Handset

- Dterm PSIII

Dterm IP Soft Phones

- Dterm SP20
- Dterm SP30

Dterm Extenders

- Dterm Extender (Analog)

Analog Terminals

NEC offers three distinct analog terminals.

Descriptions of the Analog Terminals

Description	Remarks
DTR-1-1 (WH) TEL DTR-1-1 (BK) TEL	Fully modular with Redial key, Flash key, Message Waiting lamp, Data Jack and Ring/Handset Receive Volume.
DTR-1HM-1 (WH) TEL DTR-1HM-1 (BK) TEL	Fully modular with Redial key, 'Flash' key, Message Waiting Lamp, Data Jack, eight programmable Feature/Speed Dial keys and Ring/Handset Receive Volume.
DTH-1-1 (BK) TEL	Fully modular with Redial key, 'Flash' key, Mute Key, Message Waiting Lamp, Data Jack, four programmable Feature/Speed Dial keys and Ring/Handset Receive Volume

Line Conditions of the Analog Terminals

Terminal Type	Card Type	Cable Length* (Cable 0.5/24 AWG)
DTR-1-1 DTR-1HM-1 DTH-1-1	PN-4LCD-A (max. 600 ohms loop resistance)	Approximately: 1.43km (.88 miles)
	PN-8LCAA (max. 600 ohms loop resistance)	Approximately: 1.43km (.88 miles)
	PN-AUCA (max. 2500 ohms loop resistance)	Approximately 12.29km (7.63 miles)
	PN-4LLCB (max. 2500 ohms loop resistance)	Approximately 12.29km (7.63 miles)

Note: **Cable length is based on the diameter of the cable and the terminal impedance*.

Specification for Analog Terminals

Item	Description
Size	224mm x 165mm +/5mm(Length x Width)
Color	Black or White (DTH-1-1 black only)
Dial Pad	12-Key Dial Pad: 4 Rows and 3 Columns; Metropolitan Dial Pad with Alphabet, * and # buttons; Button 5 has a Raised Dot
Type of Dial	DTMF and Dial Pulse
Function Buttons	Hook flash, Redial Key, Speaker Monitor and Hold Key(DTR-1HM Only)
Message Waiting Lamp	Neon Lamp with Window Design -Glow Through Filter Raised from Surface with MW and Incoming Ring Indication
Message Waiting Lamp Operating Voltage	Activation Voltage 88V to 108V, Deactivation Voltage 53V or Less
Speed Dials	(DTR-1HM Only) 8 Buttons (2 Columns), Maximum 21 Digits (DTH-1-1) 4 buttons, Maximum 21 digits
Hook flash Timer	630+/10ms (Fixed)
Redial Key	Maximum of 31 digits
Ring Vol. Control	4 Levels (Off Low Middle High) Programmable
Ring Tone Pitch Control	3 Levels Programmable
Handset Receiver & Speaker Volume	6 Levels (Volume Key)
Handset	Hearing Aid Compatible, Dynamic Type Element
Handset Cord	12 feet
Directory Card	Large Convenient Directory Card
Data Jack	Dedicated Jack; Used for Connection to Modem
Data Jack Position	Back of Telephone
Wall-Mount Unit	Built-in
Electrostatic Discharge	Can Withstand +/- 20kv Discharge
Approvals	c-UL (UI 60950 3rd Edition) FCC part 15, 68 IC

Dterm Series i (IP) Terminals

Dterm IP terminals offer features and benefits that brilliantly capture the essence of form and functions, helping your business take service and performance to newer, more profitable heights. The Dterm IP terminals consist of four distinct models, a 4, 8, 16 or 32 feature key terminals, in order to meet diverse telephone terminal needs.

Descriptions of the Dterm IP Terminals

Description	Remarks	Physical Features
ITR-4D-1 (BK)	4 Programmable Line/Feature Keys with Display	• Tilt-able LCD Unit • Adjustable Base • Built-in Wall Mount Unit
ITR-8D-3 (BK) ITR-8D-3 (WH) ITR-8D-3 (BB)	8 Programmable Line/Feature keys with Display	
ITR-16D-3(BK) ITR-16D-3(WH) ITR-16D-3(BB)	16 Programmable Line/Feature keys with Display	
ITR-16LD-3(BK) ITR-16LD-3(WH)	16 Programmable DESI-LESS Line/Feature keys with Display	
ITR-32D-3(BK) ITR-32D-3(WH) ITR-32D-3(BBK)	32 Programmable Line/Feature keys with Display	

Call Handling Features

- Four Local Soft Key Controls (detail functions are dependent on PBX)
- Large Message Waiting LED
- 24 Character, 3-Line LCD
- Built-in Headset Jack Connector **Note**
- Speed Dial/DSS Buttons
- Programmable Line Keys with 2-Color LED
- Eleven Dedicated Function Keys (Feature, Recall, Conf, Redial, Hold, Transfer, Answer, Speaker, MIC, Directory and Message **Note**)
- Built-in Half Duplex Hands-free Unit

Note: *DTR-4D-1 does not include headset jack or directory and message functions*

Convergence Features

- Two 10/100 full duplex Ethernet ports- One which connects the Dterm IP to the local Ethernet Network, the other provides connectivity for a local workstation.

- Three types of powering options:
  - Local AC adapter
  - CDP (Cisco Discovery Protocol) power for those infrastructures with an installed base of Cisco gear
  - Spare pair power across the Ethernet network.

- Transportable QoS, which follows the user no matter where they log in.
  - Multiple Voice Coding support, which automatically negotiates to a common setting.
  - G.711 providing an international standard for encoding/decoding telephony signals on a 64 Kbps Non-compressed channel.
  - It also supports the compression algorithms G.729A (8Kbps) and G.723.1 (5.3/6.3 Kbps).

Line Conditions

ITR-4D-1	Max. 328 ft. between Ether/IP-PAD and Router/ Switching Hub
ITR-8D-3	
ITR-16D-3	
ITR-16LD-3	
ITR-32D-3	

Dterm Series i (TDM) Multi-line Digital Terminals

The Dterm Series i, digital terminals have a user-friendly ergonomic design and is available in six distinct digital models.

Descriptions of Terminals

Description	Remarks
DTR-2DT-1 (BK) TEL DTR-2DT-1 (WH) TEL	2 LINE TERMINAL - available in black (BK) or white (WH). (does not support optional adapters) Fully modular with 2 Flexible, 2-color LED Line keys, eight Function Keys, built-in Speakerphone and electronic volume and tone controls.
DTR-4D-1(BK) TEL	4 LINE TERMINAL – available in black only. Fully modular with 4 Flexible, 2-color LED Line keys, nine Function Keys, built-in Speakerphone, wall mount unit, four soft keys, Large LED, Electronic Volume and Tone Controls and tilt stand. (does not support optional adapters)
DTR-8-2 (BK) TEL DTR-8-2 (WH)TEL	8 LINE TERMINAL – available in black (BK) or white (WH). Fully modular with 8 Flexible, 2-color LED Line keys, eleven Function Keys, built-in Speakerphone, headset jack, wall mount unit, four soft keys, Large LED, Electronic Volume and Tone Controls and tilt stand.
DTR-8D-2 (BK) TEL DTR-8D-2 (WH) TEL	8 LINE DISPLAY TERMINAL - available in black (BK) or white (WH). Fully modular with 8 Flexible, 2-color LED Line keys, eleven Function Keys, built-in Speakerphone, headset jack, wall mount unit, 24-character by 3-line display, four soft keys, Large LED, Microphone button, Message button, Directory button, Electronic Volume and Tone Controls, and tilt stand.
DTR-16D(BL)-2(BK) TEL	16 LINE BACK-LIT DISPLAY TERMINAL – available in black only, 16 flexible 2 color line keys, eleven function keys, built-in speaker phone, headset jack, wall mount unit, back-lit 24-character 3 line display, four soft keys, large LED, electronic volume and tone controls and tilt stand
DTR-16D-2 (BK) TEL DTR-16D-2 (WH) TEL	16 LINE DISPLAY TERMINAL - available in black (BK) or white (WH). Fully modular with 16 Flexible, 2-color LED Line keys, eleven Function Keys, built-in Speakerphone, headset jack, wall mount unit, 24-character by 3-line display, four soft keys, Large LED, Microphone button, Message button, Directory button, Electronic Volume and Tone Controls and tilt stand.
DTR-16LD-1 (BK) TEL DTR-16LD-1 (WH) TEL	16 LCD DESI-LESS TERMINAL – 16 Flexible LCD appearances for Speed Dial/DSS, feature or line appearance, 11 Function Keys, Built-in Speakerphone (Half Duplex), Headset Jack, Wall Mount Unit, 4 Soft-keys, In-Line Power and 12 ft. Handset Cord
DTR-32D-1 (BK) TEL DTR-32D-1 (WH) TEL	32 LINE DISPLAY TERMINAL - available in black (BK) or white (WH). Fully modular with 32 Flexible, 2-color LED Line keys, eleven Function Keys, built-in Speakerphone, headset jack, wall mount unit, 24-character by 3-line display, four soft keys, Large LED, Electronic Volume and Tone Controls and tilt stand.
DCR-60-1 (BK) CONSOLE DCR-60-1 (WH) CONSOLE	ATTENDANT ADD-ON CONSOLE - Requires an AC-R ADP (included). Fully modular with 48 programmable, 2-color LED keys (for station trunk appearances), 12 Function keys with red LED, and tilt stand. Terminal available in: black (BK) or white (WH).
D16(LD)-R ADM (BK) D16(LD)-R ADM (WH)	16 LCD DESI-LESS ADD ON MODULE – available in black (BK) or white (WH) Provides Speed dial directory and one-touch speed dial.

Standard features

- Adjustable Base
- Built-in Wall Mount Unit
- Four Local Soft Key Controls (detail functions are dependent on PBX, only provided on terminals with display)
- Built-in Half Duplex Hands-free Unit
- Large Message Waiting LED
- 24 Character, 3-Line LCD on display equipped models
- Built-in Headset Jack Connector **Note**
- Speed Dial/DSS Buttons
- Programmable Line Keys with 2-Color LED
- Eleven Dedicated Function Keys (Feature, Recall, Conf, Redial, Hold, Transfer, Answer, Speaker MIC, Directory and Message **Note**)
- Ringing Sound Variation
- LCD indication of volume control
- FCC Part.15 Class B Approved
- Replaceable Function/Dial Key pad:
  - Used for foreign languages, ACD instruments, etc.
- Un-tangler compatible handset
- The Directory card can swivel 360°

Note: *DTR-4D-1 does not include headset jack or directory and message functions*

Specification

Item	Description
Curl Cord Length	12ft
Weight (no handset)	510g (min.)
LCD Display	24 digit x 3 line (alphanumeric and some characters) no back light
Ringing Sound Level	Max. 80dBSPL (in output limit condition) max. 86dBSPL (in normal condition)
Built in Hands Free	Half duplex
Items Provided with Instrument	Line cord, Directory card
Handset Cradle	K type compatible
LCD angle	14~42.5 deg. (on the desk, no housing tilt) 25~53.5 deg. (on the desk, housing tilt up) -4.4 deg. (wall mounting)
Housing Face Angle	14~25 deg. (on the desk) -4.4 deg. (wall mounting)
Recommended Headset	NEC Headsets
Other	HAC

Terminal Options

Item	Description
WM-R Unit	Series i Multi-line Terminals with an AP (R)-R, AP (A)-R, CT (A)-R, and/or an IP-R Unit can be wall mounted using the WM-R Unit.
AD(A)-R Unit	Provides Series i Multi-line Terminals ability to interface with recording device.
AP(R)-R Unit	Provides Series i Multi-line Terminals ability to interface with analog device such as a cordless telephone, facsimile machine, external speakerphone, Automatic Dialer or modem. Provides ringing to analog device connected. Requires an AC-R ADP.
AP(A)-R Unit	Provides Series i Multi-line Terminals ability to interface with analog device such as a cordless telephone, facsimile machine, external speakerphone, Automatic Dialer or modem. No ringing is provided.
CT(A)-R Unit	Connects a Series i Multi-line Terminal to a PC providing a complete turnkey package with graphical telephone user interface and call logging. Shipped with Multi-line Phone Kits software. Supports Serial interface.
IP-R Unit	A compact plug-and-play device that installs into the base of a Dterm Series i display terminal. Integrated two-port 10/100baseT Ethernet Pass through hub that permits using one port to connect the network interface card (NIC) from the PC to the IP network. The other is plugged directly into a LAN or an IP network device such as a router, DSL modem or cable modem. Requires an AC-R ADP.
AC-R Unit	AC adapter for DSS/BLF Console, AP(A), AP(R), or IP-R Unit

Line Conditions

Cable Length Note 1			Standard	with AC Adapter
Dterm Series i	Dterm 4D Dterm 8 / 8D Dterm 16/16D/16LD	8DLC	300m (984ft)	**Note 2**
		4DLC	300m (984ft)	1200m (3937ft)**Note 4**
		2DLC	850m (2789ft)	1200m (3937ft)**Note 4**
	Dterm 32/32D	8DLC	200m (656ft)	**Note 2**
		4DLC	200m (656ft)	1200m (3937ft)
		2DLC	850m (2789ft)	1200m (3937ft)
	DSS/BLF Console **Note 3**	8DLC	-	300m (984ft)
		4DLC	-	1200m (3937ft)
		2DLC	-	1200m (3937ft)

Note 1: *Cable length is based on the following conditions.*
- *Diameter of the cable is 0.5 mm.*
- *The Protection arrester is not inserted between the terminal and PBX.*

Note 2: *When using 8DLC card, long line function is not availability, even if it is equipped with AC Adapter.*

Note 3: *The DSS/BLF Console requires local AC/DC supply.*

Note 4: *The DTR 4D-1 does not support an AC adapter.*

Dterm Series E (Multi-line Digital Terminals)

The Dterm Series E terminals were strategically designed to provide ergonomic form and user-friendly functions. With advanced digital circuitry, the Dterm Series E terminals consist of five distinct telephone models and one DSS/BLF console to meet users' diverse terminal needs. The Dterm Series E family of terminals offers adjustable display and non-display units with menu-driven soft key operation, allowing users to program terminals at the desktop. The display units are equipped with large LCD panels with three lines of display, each with 24 characters. Each terminal offers an optional full duplex speaker phone operation for two-way conversation. Standard features include headset jacks, wall mount units and adjustable base units. Laser printer designation cards are available in Metallic Green (MG), Black (BK) and Soft White (WH).

Standard features

- HOLD – Press key to place an internal or external call on hold.
- TRANSFER – Allows the station user to transfer established calls to another station without attendant assistance.
- ANSWER – When LED on this key is lighted, press key to answer a waiting call.
- SPEAKER – Controls the built-in speaker that can be used for Hands Free Dialing/Monitoring and Voice Call.
- REDIAL – Press key to activate redial feature. Press redials and scrolls back through numbers that have been dialed. When desired number is displayed, press the * key to activate dialing.
- CONF – Press key to establish a three-way conversation.
- RECALL – Press key to terminate established call and re-seize internal dial tone.
- FEATURE – Used to activate terminal setup functions and to program One-Touch Speed Dial/Feature Keys.
- MIC – Press to turn on and off the microphone.

Dterm Series E Soft keys

The Dterm Series E Display Terminals have four Soft keys located just under the Display of each Terminal. These menu-driven Soft keys allow user's convenient access too many features. Each Soft key may have up to 4 features programmed on it. Soft keys will be functional for specific states of the terminal such as idle, during dialing, ringing, etc. The state of the terminal will determine what Soft key is available to the user. According to the status of the Multiline Terminal, functions of the Soft keys are displayed in the third line on the LCD and if the status of the Multiline Terminal changes, the Soft Key displayed will change automatically. Also if the HELP key is pressed, an explanation of the indicated Soft key is shown on the LCD. Typical Soft key Applications are Business, Hotel, Multi-tenant and Networking applications where it is necessary to save key space on the telephone and also provide a variety of quickly accessible features.

Descriptions of Series E Multi-line Digital Terminals

Description	Remarks
DTP-2DT – 1 (WH)	This 2-line non-display digital terminal is equipped with 2 programmable line/feature keys (each with a two-color LED), 8 dedicated function keys, a built-in speakerphone, and large LED to indicate incoming calls and messages. The terminal comes in Ivory White (WH) only.
DTP-8 – 1 (WH) DTP-8 – 1 (BK)	This 8-line non-display digital terminal is fully modular with 8 dedicated function keys, 8 programmable line/feature keys (each with a two-color LED), built-in speakerphone, built-in headset jack, built-in long loop adapter, built-in wall mount unit, tone/volume/contrast control, tilt stand, and a large LED to indicate incoming calls and messages. The terminal comes in two colors: Black (BK) and Ivory White (WH).
DTP-8D – 1(WH) DTP-8D – 1(BK)	This 8-line display digital terminal is equipped with 8 dedicated function keys, 8 Programmable line/feature keys (each with a two-color LED), built-in speakerphone, built-in headset jack, built-in long loop adapter, built-in wall mount unit, tone/volume/contrast control, tilt stand, 3 line by 24 character LCD display panel (adjustable/detachable), four soft keys, and a Large LED to indicate incoming calls and messages. This terminal comes in two colors: Black (BK) and Ivory White (WH).
DTP-16D – 1 (WH) DTP-16D – 1 (BK)	This 16-line display digital terminal is equipped with 8 dedicated function keys, 16 Programmable line/feature keys (each with a two-color LED), built-in speakerphone, built-in headset jack, built-in long loop adapter, built-in wall mount unit, tone/volume/contrast control, tilt stand, 3 line by 24 character LCD display panel (adjustable/detachable), four soft keys, and a Large LED to indicate incoming calls and messages. This terminal comes in two colors: Black (BK) and Ivory White (WH).
DTP-32D – 1 (WH) DTP-32D – 1 (BK)	This 32-line digital terminal is equipped with 8 dedicated function keys, 32 Programmable line/feature keys (each with a two-color LED), built-in speakerphone, built-in headset jack, built-in long loop adapter, built-in wall mount unit, tone/volume/contrast control, tilt stand, 3 line by 24 character LCD display panel (adjustable/detachable), four soft keys, and a Large LED to indicate incoming calls and messages. This terminal comes in two colors: Black (BK) and Ivory White (WH).
DTP-32DA-1 (WH) DTP-32DA-1 (BK)	See above for basic description of phone. This phone is to be used as an ACD terminal. The Silk Screening of the function keys is changed for the dedicated ACD functions.

Description of Series E Multi-line Digital Terminals (Cont)

Description	Remarks
DCU-60 –1 (WH) DCU-60 –1 (BK)	This DSS/BLF/Add-On Console is equipped with 60 programmable line keys (each with a two-color LED). These keys can be programmed as Direct Station keys, function keys, or outside line keys. This terminal comes in two colors: Black (BK) and Ivory White (WH). When the console is used as an Add-On Module, an additional 25 lines/trunks and the existing line/trunks for the Multiline Terminal can be accessed directly (maximum 41 lines/trunks). The station speed dialing function can be assigned for all keys on the Add-On Module unit. Also, one of the last three keys can be used as a Day/Night change keys. When the console is used as a DSS/BLF console, up to 60 stations can be directly accessed in addition to those already appearing on the Multiline Terminal. A red LED associated with each button indicates busy status for each station. In addition, the DSS/BLF console can provide the following functions: Message Waiting – Set/Cancel/Status Display Do Not Disturb – Set/Cancel/Status Display Automatic Wake Up No Answer – Status Display/Cancel Agent Busy Out – UCD – Status Display Line Lockout – Status Display Room Cutoff – Set/Cancel/Status

Dterm Series E Accessories & Adapters

Unit	Description
ADA-U	Ancillary Device Adapter (ADA-U) The Ancillary Device Adapter (ADA-U) allows for a direct connection to a tape recorder for logging/recording telephone calls. A dedicated set of input connectors is also provided for a recording tone unit to inform the parties that the call is being recorded. The ADA-U does not require an AC adapter.
HFU-U	External Hands free Unit (HFU-U) The External Hands free Unit (HFU-U) provides a solution for small office teleconferencing by improving the sound quality of speakerphone calls via an external microphone. This unit is ideal for a working environment where hands free calling is a necessity. In order to provide maximum performance, two user-adjustable switches are provided which allow the speakerphone to be configured for a unique environment (quiet room, noisy business environment, and room with acoustic echo). Furthermore, a push-to-mute button is featured on the external microphone to add privacy for hands free calling.
APR-U	Analog Port Adapter with Ringing (APR-U) The Analog Port Adapter with Ringing (APR-U) provides an interface for installing single line telephones (SLTs), modems*, and the NEC VoicePoint/VoicePoint + conferencing unit. By installing the APR-U, calls may be initiated via a PC modem and switched to voice operation if desired. Data calls at a maximum speed of 28.8 kbps are also supported with the APR-U. The APR-U has the added benefit of detecting incoming ringing signals. By providing ring detection, the user may wish to install a personal fax machine or an answering machine for convenience. Finally, two user-adjustable switches are provided on the adapter: one allows for either a 600 □ or complex impedance interface to a (modem, SLT, etc.), also a second voice channel to permit calls from the Multiline Terminal and devices installed to the APR-U simultaneously*. The APR-U requires an AC adapter. **Note 1*:** *When a modem is installed to the Multiline Terminal with the APR-U, the modem and the Multiline Terminal (handset, speakerphone, and headset) cannot be used simultaneously.* **Note 2*:** *Dual voice channel capability is available and special programming may be required from the NEC PBX/KTS systems to support two simultaneous voice channels.*
CTA-U	Computer TAPI ADAPTER Unit (CTA-U) The Computer TAPI Adapter Unit (CTA-U) permits connecting a Dterm Series E multilane terminal to a PC. Each CTA-U unit is shipped with Multiline Phone Kits software, a complete turnkey package offering a graphical telephone user interface and call logging.

Dterm Series E Accessories & Adapters (Cont)

ACA-U	AC Adapter Unit (ACA-U) The AC adapter provides the Dterm Series E terminal with the voltage required to power the built-in long loop adapter, and APR units. One AC adapter can power up to three optional adapters.
WMU-U	External Wall Mount Unit (WMU-U) The External Wall mount Unit is ONLY required if an option adapter is required on the Multiline terminal which is being mounted on a wall. It permits the terminal to be installed to a flat wall surface. With this unit, any option adapter may be installed to the terminal for high functionality. The built-in wall mount unit should be used for wall mount operation when no option adapter (except ADA-U unit) is installed on the Multiline terminal. **Note:** *This unit is designed to enhance the hands free operation of Multiline Terminal by providing an echo canceling circuit. However, this unit is primarily designed for a typical small office environment and not for conference rooms. Its performance should not be compared to commercial, expensive audio conference units. The HFU-U requires an AC adapter.*
WMU-W	Wall Mount Unit for DTP-2DT-1 terminal
PC Telephony Adapter	PC Telephony Adapter The PC Telephony Adapter provides an integration link from a desktop PC to the Dterm Series E telephone. Using Microsoft® Windows™ Telephony API, Windows applications users can now control or use many of the features, desktop message notification, calling line identification, and calling functions of the Dterm Series E telephone from their PC.
IPW-2U (ELC)	IP Adapter for Dterm Series E. For use with 32 Button Display Terminal and PN-32IPLA ELC station card only. Requires either ACA-U local power or SN1604 Power Hub 12-Port
IPW-2U (P-P)	The IPW-2U (P-P) adapter is a full duplex switch, which can be installed on any of the NEC LCD Dterm Series E phones when using Peer-to-Peer IP stations. Requires either ACA-U local power or SN1604 Power Hub 12-Port

Dterm Series E Specifications

Feature/Function	Dterm Series E Digital Terminals (DTP-)					DCU-60	
	2DT	8	8D	16D	32D	Add-On	DSS
Housing color	White	White or Black					
Two colors LED	Yes	Yes	Yes	Yes	Yes	No	No
Line/Feature keys	2	8	8	16	16	48 ext.	60
One touch keys	None	None	None	None	16	12	None
Fixed feature keys	8	8	8	8	8	None	None
Number of characters per line on display	N/A	N/A	24	24	24	N/A	N/A
Number of lines on display	N/A	N/A	3	3	3	N/A	N/A
Volume Control							
Speakerphone	Yes	Yes	Yes	Yes	Yes		
Headset	No	Yes	Yes	Yes	Yes		
Ring Volume	Yes	Yes	Yes	Yes	Yes	N/A	N/A
LCD Contrast	N/A	N/A	Yes	Yes	Yes		
Soft keys	N/A	N/A	Yes	Yes	Yes	N/A	N/A
Off hook ring on/off key	No	No	Yes	Yes	Yes	N/A	N/A
Mute on/off key	No	No	Yes	Yes	Yes	N/A	N/A
Microphone on/off key	No	No	Yes	Yes	Yes	N/A	N/A
Headset on/off key	No	No	Yes	Yes	Yes	N/A	N/A
Built-in speakerphone	Yes	Yes	Yes	Yes	Yes	N/A	N/A
Built-in headset jack	No	Yes	Yes	Yes	Yes	N/A	N/A
Built-in wall mount unit	Optional	Yes	Yes	Yes	Yes	Yes	Yes
AC adapter required	Optional	Optional	Optional	Optional	Optional	Yes	Yes
Data Adapter	No	No	No	No	No	N/A	N/A
Message Waiting Lamp	Yes	Yes	Yes	Yes	Yes	N/A	N/A
Add-on console supported	Yes	Yes	Yes	Yes	Yes	N/A	N/A
ADA supported	No	Yes	Yes	Yes	Yes	N/A	N/A
APR supported	No	Yes	Yes	Yes	Yes	N/A	N/A
CTA supported	No	Yes	Yes	Yes	Yes	N/A	N/A
PC Telephony Adapter supported	No	Yes	Yes	Yes	Yes	N/A	N/A
HFU supported	No	Yes	Yes	Yes	Yes	N/A	N/A
HFU operation:							
Half Duplex	Yes	Yes	Yes	Yes	Yes	N/A	N/A
Full Duplex	No	Yes	Yes	Yes	Yes		
Adjustable legs	No	Yes	Yes	Yes	Yes	Yes	Yes
Long loop supported	Yes	Yes	Yes	Yes	Yes	N/A	N/A
Hearing Aid compatible	Yes	Yes	Yes	Yes	Yes	N/A	N/A
Cable pairs	1	1	1	1	1	1 port	2 ports

Dterm Series E Line Conditions

Loop Resistance and Distance (Long Loop circuit built-in)				
Line Card	Terminal	Maximum Loop Resistance (□)	Cable Length (24 AWG Cable) (Without ACA Unit)	Cable Length (24 AWG Cable) (With ACA Unit)
4DLC	DTP-8-1 TEL	35	1000 ft. (304 m)	4000 ft. (1219 m)
	DTP-8D-1 TEL	35	1000 ft. (304 m)	4000 ft. (1219 m)
	DTP-16D-1 TEL	26	650 ft. (198 m)	4000 ft. (1219 m)
	DTP-32D-1 TEL	21	650 ft. (198 m)	4000 ft. (1219 m)
	DCU-60-1 Console	N/A	N/A	1000 ft. (304 m)
8DLC	DTP-8-1 TEL	35	1000 ft. (304 m)	1000 ft. (304 m)
	DTP-8D-1 TEL	35	1000 ft. (304 m)	1000 ft. (304 m)
	DTP-16D-1 TEL	26	650 ft. (198 m)	1000 ft. (304 m)
	DTP-32D-1 TEL	21	650 ft. (198 m)	1000 ft. (304 m)
	DCU-60-1 Console	N/A	N/A	1000 ft. (304 m)
2DLCB/N	DTP-8-1 TEL	35	2800 ft. (853 m)	4000 ft. (1219 m)
	DTP-8D-1 TEL	35	2800 ft. (853 m)	4000 ft. (1219 m)
	DTP-16D-1 TEL	26	2800 ft. (853 m)	4000 ft. (1219 m)
	DTP-32D-1 TEL	21	2800 ft. (853 m)	4000 ft. (1219 m)
	DCU-60-1 Console	N/A	N/A	2800 ft. (853 m)

Dterm Cordless Terminals

Dterm Cordless II (DTR-4R-2 BK)

The Dterm Cordless II is fully digital and operates in the 900MHz frequency range. It includes Base Unit, Handset (with a standard battery, 6 hours of Talk Time or 4 days of Standby), Charging Stand, Belt Clip, Wall Plate Adapter and 2 AC Adapters.

Specifications:

- Digital
- Up to 350 feet
- 900 MHz digital spread spectrum
- 4 programmable line keys
- 10 Simultaneous channel access
- Headset connection
- Volume slide switch/6 ring tones
- Weight: 8.8 ounces w/battery

Options:

Unit	Description
EXP 9530	Headset Lightweight headset with foam covered ear and microphone Mounts over the head
EXP 9785	REPLACEMENT CHARGER UNIT w/o AC Adapter The charging cradle is used to charge the batteries used with the Dterm Cordless II Terminal
EXP 9702	REPLACEMENT SPRING-TYPE BELT CLIP Use to clip the handset to clothes or belt
AD 970	REPLACEMENT AC ADAPTER FOR CHARGING UNIT Provides 9Vdc to the Dterm Cordless II Charging Unit Requires 120V, 60Hz (14W) AC input from standard AC plug
EXP 9704	REPLACEMENT AC ADAPTER FOR BASE UNIT Provides 10Vdc to the Dterm Cordless II Terminal Unit Requires 120V, 60Hz (14W) AC input from standard AC plug
Leather Carry Case	LEATHER CARRY CASE Leather case for DTR-4R-1 (BK) Handset
Wall Mount	Replacement Wall Mount Adapter Used to mount the Dterm Cordless II to a standard wall plate
BT 2499	REPLACEMENT BATTERY Dterm Cordless II - 6 hours of Talk Time or 4 days of Standby Capacity

Dterm Cordless Lite II (DTH-4R-1 BK)

The Dterm Cordless Lite II is digital and operates in the 900 MHz narrow band frequency range. It includes Base Unit, Handset, one telephone cord, 2 AC Adapters, 2 Rechargeable Batteries, Spring Belt Clip, Battery Charger, and Wall Mounting Plate.

Specifications	
Frequency	900Mhz FM with ADPCM
Security	Digital
Voice Channel	30 Channel
Interface Port	Digital/Analog
Multi Line Capability (Hand Set)	4 Line Capability
LCD Screen	2x16 Digit (10 Digit)
System Caller ID	YES
Remote Base Unit	YES
2nd Battery Charging system	NO
Auto Talk	YES
Auto Standby	YES
Talk Time	6 Hours
Standby Time	96 Hours
Silent Alert System	YES
Handset Size	5 ½
Ring On-Off Key	YES
Battery Type	NiMH Battery (3.6V-700mAh)
Selectable Ring Tones	3 Tone
Message Waiting Indicator	YES
Belt Clip	YES
Headset Jack	Yes-2.5mmö
Transfer	YES
Conference	YES
Hold	YES
Mute	YES
Backlit Keypad	4 LINE KEY
Hot Swap Battery	YES

Options

Unit	Description
EXP9660	Replacement Wall Mount Unit Use to wall mounts the Dterm Cordless Lite Terminal.
AD9500	Replacement AC Adapter for Charging Base Unit.
AD960	Replacement AC Adapter for Charging Unit. Provides 9Vdc to the Dterm Cordless Lite Charging Unit. Requires 120V, 60Hz (14W) AC input from standard AC plug.
BT9000B	Replacement Battery 5 hours of Talk Time or 40 hours of Standby Capacity, 500mAH 4.0V
EXP9603	Replacement Spring-Type Belt Clip. Use to clip the handset to cloths or belt.
EXP9683	Leather Carry Case.
EXP9685	Replacement Charging Cradle and AC Adapter.
EXP9530	Dterm Headset. Lightweight headset with foam covered earpiece and microphone.
EARSET	Dterm EARSET. Lightweight ear set with microphone.

Plantronics CT Cordless

Package includes Base Unit, Handset (with a standard battery 6 hours of Talk Time or 3½ days of Standby), Belt Clip, M175 Headset, User Guide and AC Adapter. This compact unit is an Analog single line device that operates in the 2.4 GHz Digital Spread Spectrum frequency range. This unit is compatible with and Analog single line port, Analog Port Adapter or Analog telephone line from Telco.

Specifications

- 2.4 GHz Cordless Headset Phone
- Range of Up To 150 Feet
- Audible Low Battery Indicator
- Single Line Operation
- Ultra-Compact Remote Unit with Belt Clip
- Variable Range Volume Control
- 10 Number Speed Dial
- Page / Find Feature
- Redial / Flash
- Mute with Audible Reminder
- Talk / Charge / Power Indicator Lights
- Built In Headset Stand

INASET

INASET terminals are members of the Dterm IP family. INASET terminals have a Web browser with a large color display and a built-in multi-port Ethernet switch for connectivity to the user's local PC. INASET terminals bring a wealth of information to the desktop, including short text display messages and Web pages specifically tailored for the small screen format

The INASET's basic load includes a graphical telephony application that provides an abundance of telephony information and desktop control that is easy to use with its menu-based interface. Information and controls accessible via the soft keys and feature buttons include:

- **Line status** showing a visual icon display for the status of all assigned phone lines and DSS lines.
- **Caller information** showing a visual text display for things such as time, date and call status information.
- **Telephony Directory** for storing, searching and dialing different profiles which you can categorize and store in one of three different groups: corporate, personal and group.
- **Web access** providing browsing capabilities to display HTML web-based information located on the Internet or Intranet. Also includes support for Java applets.
- **Virtual keys** providing access to features, functions and recent keys activated on the terminal. The user can program display and functions for how they see fit with limited or no administrative support necessary.

By converging a company's voice and data networks, there is only one network to manage. The INASET includes a built-in switch, so you can use a single Ethernet switch port for the computer (data) and the INASET. Because it's an IP based telephone, it can be installed anywhere on a corporate IP network. The INASET is Dynamic Host Configuration Protocol (DHCP) compatible and doesn't require the need to be co-located with the NEAX PBX equipment. INASET supports G.711, G.729a and G.723.1 audio compression for low-bandwidth requirements.

INASET Applications

The INASET is specifically suited for the enterprise environment, including the following end users: managers, purchasing agents, consultants and call agents. But this is just the beginning. This advanced IP business terminal can be programmed to do much more than a standard business phone. For example, the customizable user interface (that's developed with the SDK) can replace single line phones that are typically installed in cafeterias, break rooms, lobbies and manufacturing floors. The INASET's programmability is also ideal for traders, stockbrokers, real estate brokers, executives, PR professionals and any other occupation in which professionals use Internet-related information to interact with the public.

Additional Applications

- **Personalization at the desktop:** Individual customization can bring flexibility to the desktop. Software developers can enhance traditional features such as Answer, Redial, Conference, Recall and Help menus.

- **Centralized services:** Information or features that need to be accessed by numerous individuals within the organization can be centralized on servers. Directory service functions, Intranet web site information, customer records, CTI applications and other work group

applications are examples of centralized services that could improve portability and use throughout the enterprise network.

INASET Terminal Line UP.

The INASET terminals are available in three models. Each is described below.

Description	Remarks
ITR-240G-1 (BK)	The INASET 240G has a 3" LCD with a gray scale 240 x 160 pixel display and 16 programmable keys. Supports Power over Ethernet (POE) (802.3af and Cisco Discovery Protocol). Available in black only.
ITR-320C-1 (BK)	The INASET 320C has a 5.1" color display and 32 programmable keys. Supports Power over Ethernet (POE) (802.3af and Cisco Discovery Protocol). Available in black only.
ITR-320G-1 (BK)	The INASET 320G has a 5.1" LCD with gray scale display and 32 programmable keys. Supports Power over Ethernet (POE) (802.3af and Cisco Discovery Protocol). Available in black only

INASET Accessories

Description	Remarks
WM-RL UNIT	Wall mount for the ITR-LC-1
INASET Handset Cord 12'	12 Foot Black Handset Cord
INASET Handset Cord 25'	25 Foot Black Handset Cord
AC-R	AC adapter for the INASET
ILPA-R	802.3af power Dongle for INASET Original
ADA-2R	Audio recording adapter for IP terminals
MIC-R (BK/WH) UNIT	External Microphone for INASET 240, 320s
PSA-R	Analog survivable adapter

INASET Basic Network Capability

- Dual port 10/100 Mbps Ethernet switch
- Internal voice packet prioritization
- VLAN support (802.1q & 802.1p frame tagging)
- Configurable ToS bits (DiffServ and IP Precedence support)
- Powering
  - Inline power (unused pairs) or
  - Direct powering with external wall-plug adapter

PBX System Requirements

- NEAX 2000 IPS R6.1or higher
- IP-PAD Card
- 8 Seat Licenses: Supports up to 8 INASET terminals

Dterm PS III Wireless Handset

The Dterm PSIII is a wireless handset designed to support the professional's schedule and the special needs of people on the run. At less than 3.8oz and with 300 hours of standby time, the Personal Station continues to offer the size, weight and battery life demanded by the user constantly on the go. The phone uses digital technology with audio quality approaching that of a digital desk phone. The new DtermPSIII wireless handset moves freely throughout the building with seamless handover when moving from one zone transceiver to another. **Note**

Dterm PS III Features		
• Separate Call Hold & Transfer keys	• Name Display	• Last Number Redial
• 2-Line Operation	• Directory Dial	• Automatic Answer
• Enhanced Short Text Message (STM)	• Speed Dial	• Vibration Alert
• Reminder tones for unread STM	• 4 Line 18 digit LCD Display	

Dterm PS III OPTIONS

- **Standard Charger** for the Dterm PSIII handset. Single AC adapter is included.

There are three cases available for Dterm PSIII:

- Two types of **Leather Case** provide a Glove like fit for ease of handling and are designed for indoor use. One includes a swivel belt clip with quick release and the other has a built in belt clip.

- The **Weather Case** model offers a weather resistant material for users whose job is outdoors a majority of time.

There are two available headset options:

- The **Headband** type is a classic banded headset with a microphone extension.
- The **Single Ear Cover** type is an ear bud type with an in-line microphone.

Note: *For seamless handover ZT must be ver7.2 or higher*

Dterm SP30 SoftPhone

The Dterm® SP30 allows customers to capitalize on the advantages of a converged voice and data network whether they're in the office or on the road. The Dterm SP30 combines traditional business communication needs with the data applications your customers require.

The Dterm SP30 optimally delivers high quality voice via a USB-connected headset. The Dterm SP30 has the ability to use a wireless handset (PS) or Dterm for all voice connections instead of the USB handset. With a simple drag and drop, the Dterm SP30 allows telephone dialing from other telephone directory applications such as Microsoft Outlook®, HTML pages and Word® documents, etc. In addition, the Dterm SP30 provides an interface to Microsoft's Telephony Application Programming Interface (TAPI) via NEC OpenWorX integration, allowing TAPI-enabled applications, such as Outlook and ACT, to make and receive calls. The Dterm SP30 can be displayed in 1 of 4 different colors (black, red, gold and neon blue) in order to reflect the "personality" of the day. The Dterm SP30 also allows for 3 different modes of operation.

Maximized mode: Access to full line of softphone features such as application sharing, member lists, conference mode, chatting capabilities, Internet access and many others are just one click away.

Compact Mode: L-shaped user interface, operating in a small footprint on the PC screen. Compact view allows the softphone to remain active while another application window such as a Word document; database file or email is the primary focus on the PC. With the compact view, the most popular features of the converged softphone are just a click away.

Task Mode: The softphone can be minimized and shown as a task within a Microsoft Operating System. While operating in this mode, the softphone will output an audio notification to the user upon receiving an incoming call. It will be up to the user to utilize the hot key in order to activate the Dterm SP30 application and answer the call.

ACD Mode: The Dterm SP30 ACD client can support two operation modes, ACD and Business. The Business mode is the default mode. When the ACD mode is configured a new set of function keys becomes available to the Dterm SP30 ACD user. A popup menu is also available when in the ACD terminal mode. While in ACD mode the Instant Messaging feature functions to the MY Line of the terminal, not the ACD Line.

Key Features and Benefits

The Dterm SP30 offers a full complement of station and converged features for an important segment of mobility and PC-based applications. Even though a majority of the hardware telephone features function on the softphone, the main focus for the softphone is its ability to deliver PC capabilities to the telephone.

Features:

Displays call records in Quick Reference List via the Call Log Window:
All calls are recorded in a Call Log. Telephone number, date of the call and identification as to whether or not the call was received and answered are all logged. You can find at a glance when and with whom you talked. The Dterm SP30 also offers a call memo function whereby you can record the key points of the call so that you can visually identify the individual records.

Links with PC applications:
Now you can collaborate on a white board application (Word® or Excel® document or any other application) that is operating on your PC and the PC at a distant site. Simply place a call to the far-end Dterm SP30 user and activate the application you will be collaborating on. You are now able to have a more productive conversation.

Internet Link:
The Dterm SP30 can be assigned with a common database link for fast access to a particular site. This link could be an Internet link that needs to be accessed when receiving or placing a call (i.e., you receive a call from a customer and need to look up information that is contained in a database). Previously, you needed to locate and launch the application before loading the file. Now with the Dterm SP30, simply select the Internet link and the database file is opened, bringing it all together in one user interface.

Real Time Communication:
In addition to providing a voice and data collaboration link, you can also chat with a remote Dterm SP30. This is ideal for the real quick conversation you need with a distant Dterm SP30 user. Maybe you're on a call and need to get a quick response from a co-worker. There's no need to put the customer on hold and call the co-worker for an answer. Instead, send a chat message and get your answer in real time. The Dterm SP30 will store all your chat messages in a log so that you can use them for future reference. Additional features of the Dterm SP30: forwarding control selection for different call modes, videoconferencing capabilities, automatic downloading of telephony features to the display and diagnostic capacities for audio problem notification.

Presence/Status
The presence/status functionality allows the user to confirm a buddy's presence/status with a visual indication (ICON) and text message on the Dterm SP30 GUI. The presence/status information is provided by the OpenWorX servers (LSI) package.

8/6 Party Conference Control
When the trunk conferencing card is configured within the voice switch and the voice trunk channel is configured in the program utility for the Dterm SP30, a user can dynamically setup and save future dated collaboration conferences. All the conferences can be activated immediately upon configuration or saved for a future date upon activation. When the conference is activated, the Dterm SP30, with the help of the Voice sever, places calls to the configured participants based on the number which are user predefined. Conferences do not require users to be Dterm SP30 users for voice only conference calls. Only Dterm SP30 users will utilize additional features like application sharing and messaging during call.

Collaboration
Collaboration of a Dterm or PS handset links capabilities to the Dterm SP30 GUI. In this mode, all the voice connections will be directed to a Dterm or across the PS infrastructure instead of through Dterm SP30 USB handset. The Dterm SP30 GUI will give the indication that it has received a call such as caller ID, ringing information and notification as to the state of the line. The only thing that changes is voice path being directed to the Dterm or PS handset. The Dterm SP30 will provide data collaboration interface.

Collaboration Terminal list:

- ITN Terminal (NEC SIP)
- Dterm
- Dterm IP
- Analog Terminal
- PCS/PHS

Benefits of this functionality:

- User has a reliable voice connection with the voice server.
- Dterm SP30 user has a terminal device which can be ported from place to place without the need for programming call forward functionality on the phone.
- The user can be reached by one number, no matter if call forwarding is set or the user is located next to the Dterm SP30.
- User has a physical device which they can use for dialing and answering calls instead of working with mouse and keyboards.
- User is no longer locked to their PC for receiving and answering calls. The user is as portable as the in building wireless network when using the PS.

Voice Recording

A user will be able record the voice connection and save the wave file on a network storage place of their choice. It is recommended all voice recording be stored on a local hard drive and not a network drive. With the activation of the voice recording of the Dterm SP30, no extra recording equipment other than the Dterm SP30 application with a USB headset is required. For those specific regions which require the notification to the remote party that a voice connection is being recorded, the Dterm SP30 provides for the setting of an automatic beep tone in the configuration menus. Beep tone can also be conditioned to send a tone notification at user selectable intervals.

Application Collaboration

The NEC Dterm SP30 allows users to share ideas, information and programs in a variety of ways while either in a point-to-point connection or 6/8 party conference mode.

Videoconferencing

The Dterm SP30 audio and videoconferencing feature lets you communicate with anyone on the NEC Network.

- Share ideas, information, and applications using video and audio
- Send and receive real-time images using Windows-compatible equipment
- Allows for broadcasting of the live video to other Dterm SP30 users which might not have video transmission capabilities
- Use of a video camera to instantly view items, such as hardware devices, road conditions or even personnel, which are displayed in front of the camera

Whiteboard

The whiteboard lets you collaborate in real time with other Dterm SP30 users via graphic design. With the whiteboard, you can review, create and update graphic information.

- Manipulate contents by clinking, dragging and dropping information on the white board with a mouse/keyboard.
- Copy, cut and paste information from any Windows-based application into the whiteboard.
- Use different-colored pointers to easily differentiate participant's comments.
- Save the whiteboard contents either at the local side or distant end location
- Load saved whiteboard pages, enabling you to prepare information before a conference, then drag and drop it into the whiteboard during an audio meeting

Chat

The chat functionality lets you conduct real-time conversations via text with as many Dterm SP30 users as you like. With chat, you can:

- Type text messages to communicate with other co-workers during a conference
- All messages are sent in a whisper mode so that they are only received by one party
- All messages sent and received are saved automatically in the chat log
- Automatic pop up notification when a chat message is received
- ICON notification within the chat log identifying different states of the messages

File Transfer

File transfer lets you send one or more files to distant Dterm SP30 users. With file transfer, you can:

- Send a file to other Dterm SP30 users
- Accept or reject transferred files

Application Sharing:

Dterm SP30 gives you better control over how shared programs are displayed on your desktop and give the person sharing the program control over who uses it.

- View shared programs in a frame, which makes it easy to distinguish between shared and local applications on your desktop
- Minimize the shared program frame and do other work if you don not need to work in the current conference program.
- Easily switch between shared programs using the shared programs taskbar.
- Approve conference participants' requests to work in the program you introduce.
- Allow or prevent others from working in a program using the sharing dialog menu.

Application Collaboration Modes Supported			
Applications	**2 Party Mode**	**3 Party without UM4730**	**3 Party with UM4730**
Video Conference	Yes	No	No
White Board	Yes	No	Yes
Application Sharing	Yes	No	Yes
File Transfer	Yes	No	Yes
Chat	Yes	No	Yes

One Key Operation

The space bar key on the keyboard for the Dterm SP30 PC can be configured in such a way to allow fast access to placing calls, receiving calls and termination of calls. You are no longer required to utilize a combination of the keyboard and mouse to access these telephone features.

Call Log Export

The Dterm SP30 provides for the capability of saving the call log file in a ".cvs" file format in addition to the administrator/user selecting a network storage place for the file. When opening the file with application render capable of displaying "*.cvs" files, columns breaking out caller direction, date and time, caller ID, duration of the call, notification of whether the call was recorded and memo information are displayed.

Voice Quality Alarm

The Dterm SP30 is able to provide notification to the user when it is perceived to be transmitting and/or receiving voice problems. When the Dterm SP30 application detects a voice problem an alarm is lit on the GUI and information about the audio problem is stored in the maintenance log file.

2 Line Display

When the Dterm SP30 is set for a compact mode, the display which is shown is only capable of displaying one line of information. When configuring the Dterm SP30 to a 2 line display, the display is able to switch (1 second intervals) between the normal 2 lines of display.

Automatic Idle Return

This function allows for automatic disconnection during ROT or BT tone being heard due to disconnecting by remote party. Administrator/user can choose to disable or enable this function within the user configuration. The default for this configuration is disabled.

PB tone sending
During a call, a PB tone of the dial number which is set on the member button can be sent by selecting the [send PB tone] with a right-mouse click of speed dial number.

Pause into Dialing string
This feature allows the user to insert a pause, using a comma, into the dialed number during the following conditions:

- Drag/drop into the LCD or telephone ICON.
- Copy/Paste
- Placing/transferring a call with “Call to:” tag
- Placing/transferring a call and sending the PB tone with using the member button.

Last Number Redial
A Dterm SP30 user can do a quick access and call to the most recent call placed by the SP30. During the idle stage a user can redial the telephone numbers previously dialed by selecting the [Redial] in pop up menu which appears by right-clicking on a “Call” button.

Redial via Call Log
The Dterm SP30 is capable of displaying the last 32 calling/called telephone numbers. This is done by depressing the Up/Down arrow key (Keyboard) when the LCD field is highlighted.

- First number displayed on LCD screen will be the most recent called/received telephone number.
- Pressing the arrow up will cycle from the most recent to the latest calls/received telephone number.
- Pressing the arrow down button will cycle from the last to most recent calls received/placed.

One Touch Button in Compact Mode
A button called the “One Touch” button has been added to the compact panel. The “One Touch” button allows access quick access to your speed dial list (Buddy List) that is conditioned on the main GUI under the Member buttons. By clicking the “One Touch” button, you can place a call or transfer a call to any of the member’s numbers that has pre-registered on the main GUI.

Notification of Call Received with No Answer
The Dterm SP30 Call Log button which is located on the main and compact panel will flash in blue when a call was receive by the Dterm SP30 but not answered. Before the integration of the blinking call Log button the only way a user was noted that a call was received was if the remote party left a voice mail. In those case where there was no voicemail message was left, the user had no way of identifying whether calls were received or not. Users can confirm visual that a call was received and immediately reply to the caller when they become available.

Client PC specification:

Operation Environment	
PC	IBM-PC/AT Compatible machine
OS	Windows XP/2000 w/NetMeeting application and DirectX 8.1software components installed
Memory	384Mb or more
CPU	Pentium III 900Mhz or more
Sound Device	Direct Sound Application version 8.1 or higher Full duplex capability."
Hard Drive	50Mb or more of empty capacity
LAN Interface	10/100M Ethernet
USB Port	2.0 support for USB headset for audio input/output
Video Camera	Optional

PBX Specification (Must be peer to peer IP enabled)

Operation Environment
NEAX 2000 IPS R8 Series 3300 or higher
SP30-4 software license
SP30-4 ACD software license R10 Series 3500 or higher

Dterm® SP30 Softphone Parts List

Software/Licenses/Accessories	
SP30-4 software license	Package of 4 right to use IP softphone clients.
SP30-4 ACD software License	Package of 4 right to use IP softphone w/ACD function
8 IP Seat License	Enables 8 IP phones and is required in addition to SP30 software License for SP30/SP30-ACD functionality
Dterm SP30 Software CD	Application CD. One CD can be used for multiple Client installations. Recommend 1 per 4 clients.
NEC USB Headset (DSP300)	One per SP30
NEC USB Headset (DSP400)	One per SP30

Dterm Extenders

The NEC Dterm Extenders extends the operation of the *UNIVERGE* NEAX 2000 IPS digital telephone to a remote location.

Dterm EXT (Analog)

The Dterm *Analog EXT* enables telephone users to be a fully functional part of the NEC PBX with a Dterm Series E (DTP-32DE-1) telephone located any distance off-premises. The Dterm *Analog EXT* consists of two modules. One module, identified as the Office Module, connects to a digital Dterm Series E station port on your PBX. The other module, identified as the Remote Module, connects to your NEC Dterm Series E (DTP-32DE-1) telephone at your off-premises location. The modules communicate via a single analog telephone line.

The Dterm *Analog EXT* is transparent to the user and retains access to the features and functions of the PBX. In addition, an RS-232 data port is incorporated, allowing the user to connect off-premises RS-232 equipment to equipment at the Office Module location. The Office Module emulates your telephone, and the Remote Module emulates your NEC PBX. Each module uses a V.90 internal modem (operating in V.34 mode) for the transmission of all signals between the two modules. Network configurations that support V.34 modems should work well.

Through the use of soft keys, the remote user signs on with a secured password. When the extender is online, a voice/data connection is made through public facilities via an Analog line from the remote location to the office PBX system. The Extender offers the flexibility for the user to operate in Continuous or Call-On-Demand modes. The continuous mode maintains the connection until the remote user signs off. The Call-On-Demand mode helps in avoiding continuous connection charges by only staying connected when a call is in progress.

Service Conditions for Dterm Analog EXT

- 911 calls should not be made via remote terminals since the telephone number at the PBX location is sent to the emergency center. The user should use a separate line that is not connected to the Dterm *EXT* system.
- If a dedicated line is used, it should be an unlisted number in order to minimize busy signals.
- No custom calling features, such as Call Waiting or Call Forwarding, should be ordered for the line to which you connect the Remote Module.

  Note: *A Call waiting tone causes an interruption in data transfer between the Office and Remote Module.*

- When you use voice and data simultaneously, the voice transmission will have priority over the data transmission regarding available bandwidth allocation.
- Up to 100 passwords can be programmed into each Office Module.
- Only one Remote Module at a time can be “online” with an individual Office Module.
- If passwords have been enabled on the Office Module, authorized connections require a password, 8 to 10 digits in length.
- Passwords are retained in non-volatile memory and are not affected by power outages.
- Each password begins with two digits (00-99). These digits uniquely identify up to 100 users. The 00 password is reserved for the System Administrator and can be used to change any of the user’s passwords.
- Zone paging and voice paging must be turned off via PBX programming on the ports connected to the Dterm Analog EXT. The remote telephone should be programmed to ring for all incoming calls.
- If the remote Multiline Terminal is programmed to answer multiple calls, these calls should ring at the terminal. Configure secondary incoming extensions for ringing.
- Analog fax machines and modems are not operable with the DTP-32DE-1 Multiline Terminal when connected to the APR-U optional adapter.
- The NEC Dterm Cordless and Dterm Cordless Lite telephones cannot be used in conjunction with the DTP-32DE-1 Multiline Terminal and Dterm *Analog EXT*.
- The NEC CTA-U optional adapter cannot be installed to a NEC PBX remote Multiline Terminal to provide TAPI functionally with the PBX.
- The auto answer feature for PBX ACD is not operable while in Call on Demand mode.
- The Dterm PC and Dterm PC II are not functional with the Dterm *Analog EXT*.
- The HFU-U hands free adapter is not operable with the DTP-32DE-1 Multiline Terminal for Dterm *Analog EXT* operation when used in a home environment due to FCC restrictions. However, it is permitted for use in an office environment.

Chapter 6 Trunking

Type of Trunks

The following list reflects the type of trunk and the package that supports it.

Type of Trunk	Circuit Package	LT/AP Ports	Slots
Central Office, Foreign Exchange, WATS (Analog)	PN-4COTB	4	1
	PN-8COTS	8	1
Central Office Caller ID (Class SM)	PN-4COTG	4	1
	PN-8COTQ	8	1
Direct in Dial (DID)	PN-4DIDA	4	1
	PN-AUCA	2	1
Two wire E & M tie trunks	PN-2ODTA	2	1
	PN-4ODTA	4	1
Four wire E & M tie trunks	PN-2ODTA	2	1
	PN-4ODTA	4	1
T-1 Spans	SPN-24DTAC	24	1
E-1 Spans	SPN-30DTC	30	1
Digital CCIS	SPN-24CCTA	25	1
	SPN-24DTAC & SC00	25	2
	SPN-30DTC & SC00	31	2
Analog CCIS	PN-2ODTA & SC00	2	2
	PN-4ODTA & SC00	4	2
IP CCIS (Peer to Peer)	Virtual PIM	1-128	0
IP CCIS (non Peer to Peer)	SPN-IPTB-A	4	1
	SPN-4VCTI-A	4	1-6
ISDN (PRI)	SPN-24PRTA	25	1
	SPN-24DTAC & SC01	25	2
	SPN-30DTC & SC01	31	2
BRI Trunk	SPN-BRTC	1	1
	SPN-4BRTA	4	1
Paging	PN-4COTB	4	1
	PN-8COTS	8	1
Conference	PN-CFTB	10	1
	SPN-CFTC	32	1
Announcement	PN-2DATA	2	1
	PN-4DATC	4	1
H.323 Trunk	SPN-IPTB-B	4	1
	SPN-4VCTI-B	4	1-6
Register/Sender (DTMF)	PN-8RSTG	8	1
Register/Sender (MF)	SPN-4RSTBA	4	1
Register/Sender (Caller ID)	SPN-4RSTC-A	4	1
Register/Sender (E911)	SPN-4RSTBA-911	4	1

Note: *Any combination of the above cards is allowed with a limit of 256 trunks per system.*

Trunk Parameters

- Trunk Impedance 600/900 ohms (selectable by programming)
- Trunk Resistance Characteristics, Central Office Trunk (1200 □ maximum)
- Trunk Set Up Procedure
  - Loop Start
  - Ground start
  - Wink Start
  - Ring Down
  - E & M Signaling
- Release and Incoming Detect Immediate Start and Timing Variable Start

Trunk Card Specifications

PN-4COTB

This card provides 4 Loop/Ground Start trunks with disconnect supervision, mounts in any PIM (0 to 7) and supports the following interface mixed in any combination:

- Interface for standard Analog trunks/Centrex type lines
- Interface card for CO ring down trunks
- Interface with 4 paging equipment lines
- Interface with external music/announcement source for Music-On-Hold feature

PN-8COTS

This card provides 8 Loop/Ground Start trunks with disconnect supervision and mounts in any PIM (0 to 7) and supports the following interface:

- Interface with 8 CO lines
- Interface with 8 paging equipment lines
- Interface with external music/announcement source for Music-On-Hold feature.

The above interfaces can be mixed on to one card in any combination.

PN-4COTG

This card is used to provide the system with Caller ID (Class SM). This feature requires the SPN-4RSTC card to receive the FSK signals for Caller ID. This 4-port card mounts in any PIM (0 to 7). This card supports Loop Start trunks.

PN-8COTQ

This card is used to provide the system with Caller ID (Class SM). This feature requires the SPN-4RSTC card to receive the FSK signals for Caller ID. This 8-port card mounts in any PIM (0 to 7). This card supports Loop Start trunks.

PN-4DIDA

This circuit card is a DID line trunk card which supports the functions of four DID line trunks. This card provides the following functions:

- This card is equipped with Direct Inward Dialing (DID) function.
- This card is equipped with loop detection functions, sending reverse signal, and DTMF to DP conversion.

PN-AUCA

This circuit card is used to either control two single line telephones for off-premise extensions or for 2-line Direct Inward Dialing trunks. Also, this card will connect to single telephones for Power Failure Transfer.

PN-2ODTA

The 2ODTA card is a two or four wire tie line card that mounts in any LT slot in PIM 0 to 7. This card provides analog tie line service for standard PBX networking or CCIS.

SPN-24DTAC

This circuit card is a 24 channel 1.5 Mbps digital interfacing trunk card which also supports the interface between ISDN lines and CCIS lines with 1.5 Mbps AMI. The following circuit cards may be used in conjunction with the 24DTA:

- SPN-SC01 - This circuit card controls the signal link (D channel) when Integrated Services Digital Network (ISDN) PRI (Primary Rate Interface - 23B+1D) is employed in the system. This package will support one signaling link on a single DTA package. A maximum of 8 ISDN spans are allowed per system.

- SPN-SC00 - This circuit card controls signal links when Common Channel Interoffice Signaling (CCIS) is employed in the system. This package will support one signaling link on a single DTA package. A maximum of 8 CCIS links per system is allowed

SPN-24PRTA

This circuit card is a 24 channel 1.5 Mbps digital interfacing trunk card that also provides a built-in D-Channel Handler (DCH). This package provides one DTI span with ISDN Data Link on a single package. A maximum of 8 ISDN links per system is allowed per system.

SPN-24CCTA

This circuit card is a 24 channel 1.5 Mbps digital interfacing trunk card that also provides a built-in C-Channel Handler (CCH) for CCIS. This package provides one DTI span with CCIS Data Link on a single. A maximum of 8 CCIS links per system is allowed per system.

SPN-30DTC

This circuit card is a 30 B-channel, 2 Mbps E1 digital interfacing trunk card (2.048 MHz). Typical E1 applications apply to any CCIS, Q-SIG, and TIE Line application via continuous property. Complies with U-LAW PCM voice coding standard used in North America.

Note: *Does not support A-LAW coding used in Europe, Mexico and South America.*

The following circuit cards may be used in conjunction with the 30DTC:

- SPN-SC01 - This circuit card controls the signal link (D channel) when Integrated Services Digital Network (ISDN) PRI (Primary Rate Interface - 30B+1D) is employed in the system. This package will support one signaling link on a single DTA package. A maximum of 4 ISDN spans are allowed per system.

- SPN-SC00 - This circuit card controls signal links when Common Channel Interoffice Signaling (CCIS) is employed in the system. This package will support one signaling link on a single DTA package. A maximum of 3 CCIS links per system is allowed.

SPN-BRTC

This card has one circuit of Basic Rate interface (2B+D) and provides connectivity for Voice, High Speed Data, Switched Video and G4 Fax calls to be placed and received via ISDN Basic Rate (BRI) to the ISDN Network. The BRTC card has built-in DCH (D channel handler) and has a ST interface. A locally provided NT1 device is required to connect to the ISDN Network. One BRI trunk provides two incoming/outgoing voice connections and (or) two 64k bps channels for high-speed data. This card is used for BRI trunks from the Telco to the PBX.

PN-4BRTA-F

This card has four circuits of Basic Rate interface (2B+D) and provides two 2-channel PCM digital lines. This card is used for BRI trunks from the Telco to the PBX.

PN-CFTB

The CFTB card is a 6 Party Conference card. One card can control a conference of up to ten participants. This card occupies 10 time slots per one card.

PN-CFTC

The CFTC card is a 32 Party Conference Trunk card. One card can control a conference of up to 32 participants. Two Conference methods are available: Group Call and Meet-Me Conference. This card occupies 32 AP time slots per one card.

PN-2DATA

This circuit card is a trunk card that supports a maximum of two announcements without external announcement equipment. This package supports only the following announcement arrangements:

- Two 60-second announcements.
- Single or multiple connections to an announcement is allowed.
- Announcements will be recorded from a Dterm, Single line, or Attendant Console.

PN-4DATC

This circuit card is a trunk card that supports the announcement function for a maximum of four announcements without external announcement equipment. This package supports only the following announcement arrangements:

- Four 120-second announcements.
- Single or multiple connections to an announcement is allowed.
- Announcements will be recorded from a Dterm, Single line, or Attendant Console.

PN-IPTB (IPT)

The PN-IPTB circuit card is an IP Trunk card.

- Accommodates the IP network and transmits/receives compressed voice or signals over an IP network.
- Used together with maximum of four PN-4VCTI cards (16 channels).
- 10 BASE-T/100 BASE-TX twisted pair cable is connected directly to this card.

PN-4VCTI (VCT)

The PN-4VCTI circuit card is a 4-channel CODEC card for IP Trunks; four VCT cards per IPT card.

- Voice compression protocols: G723.1, G729A, G711, FAX (14.4kbps), DTMF signals
- Used together with PN-IPTB card

PN-8RSTG (DTMF Receiver)

The RSTA card receives DTMF signals sent by single line telephones, DID lines, and tie lines. This card mounts in any LT slot in any PIM and provides 8 RST circuits per card. Usually at least one card per system is required. This card uses 8 time slots. This card is required for a 512 Port System. The 128 Port System's CPU is equipped with 4 RST circuits.

PN-4RSTBA (MF Receiver for T1 MF ANI)

This MF receiver card is required when using the T1-ANI feature. This card receives the Multi-Frequency (MF) signals sent by the CO to ID the calling party. One RSTB card provides four MF receiver ports. This card mounts in any AP slot in PIM's 0 to 7.

PN-4RSTC (ID Number Receiver)

The Caller ID (Class) feature requires this card to receive and store Caller ID information from the CO. Frequency Shift Keying (FSK) is used to send calling number and calling name information to the PBX. The RSTC card receives this information from the CO and provides flexible options for the use of the Caller ID messages.

PN-4RSTBA-911 (MF Sender)

This MF Sender card is used in conjunction with a trunk circuit to provide 911 information CAMA type interface. The trunks to be used for the E-911 System must be ordered from the local telephone company provider as E-911 trunks corresponding to the CAMA standards.

Note: *Ground Start trunks cannot be utilized for E-911 service.*

PZ-M606 (ETHERNET TCP/IP)

Peer-to-Peer IP Stations or Trunks require the PZ-M606 Ether Control Card. This card is mounted on MP card to accommodate the Ethernet transmit and receive a signal of TCP/IP protocol. 10 Base-T/100 BASE-TX twisted pair cable is connected directly to this card.

SPN-8IPLA IP PAD

The PN-8IPLA IP PAD card is an 8 channel IP to TDM Gateway with integrated G.711 CODEC (64 Kbps) and G.723.1 (5.3 Kbps), G.729a (8 Kbps) compression.

PZ-24IPLA IP PAD

The PZ-24IPLA card can be mounted to the 8IPLA card to provide 32-channels of IP-PAD with compression.

SPN-32IPLAA IP PAD

The SPN-32IPLAA IP PAD card is a 32 channel IP to TDM Gateway with integrated G.711 CODEC. Compression (G.729/G.723) requires one PN-16VCTAA IP PAD for 16 channels and two PN-16VCTAA IP PAD cards for 32 channels.

SPN-16VCTAA IP PAD

This card provides G723 and G729a vice compression for Peer-to-Peer IP Trunking.

System Trunking Capacity

IPS Trunking Capacity

Item		System Capacity							
		LPIM	2PIM	3PIM	4PIM	5PIM	6PIM	7PIM	8PIM
Central Office Trunk (Lines)	Loop Start	64	128	192	256	256	256	256	256
	DID w/4DID	48	96	144	192	240	256	256	256
Tie Line Trunk (Lines)	2W E&M	24	48	72	96	120	144	168	192
	4W E&M	24	48	72	96	120	144	168	192
CCIS Trunk (Peer to Peer Connection)		Max. 127							
DTI/CCIS Digital Link	1.5M	DTI: 10, CCIS: 8							
Note 1	2MI	8							
ISDN	1.5M/2M (PRT)	8							
	4BRT (card)	6	12	18	24	24	24	24	24
IP PAD	No. Channels	64		128		192		256	
IP Trunk		1	2	3	4	5	6	7	8
3-Party Conference		Max. 16 conference groups per system							
6-/10-Party Conference	6-Party	Max. 4 conference groups per system							
	10-Party	Max. 2 conference groups per system							
32-Party Conference		Max. 8 conference groups per system							
Announcement Trunk Circuits		128							
DTMF Sender		Max. 32 circuits per system							
DTMF Receiver		16		32					

Note 1: *The total number of trunk line and DTI channel shall be 256 or less.*

IPS DM/DML Trunk Capacity

Number of PHYSICAL MODULAR CHASSIS		Capacity Per MC	
		1	2
IP-PAD	No. of channel	32	64
Central Office Trunk (Lines)	Loop Start	56	112
	DID w/4DIDA	28	56
	2W/4W E&M	14	28
CCIS Trunk (Peer to Peer Connection)		Max. 127	
DTI/CCIS Digital Link **Note 1**	1.5M	7	DTI: 10, CCIS: 8
	2M	7	8
ISDN	1.5M/2M(PRT)	7	8
	4BRT (card)	5	10
IP Trunk		1	2
3-Party Conference		Max. 16 conference groups per system	
6-/10-Party Conference	6-Party	Max. 4 conference groups per system	
	10-Party	Max. 2 conference groups per system	
32-Party Conference		7	Max. 8 conference groups per system
DTMF Sender		Max. 32 circuits per system	
DTMF Receiver		16	32

IPS DMR Capacity

Number of PHYSICAL MODULAR CHASSIS		Capacity Per MC	
		1	2
IP-PAD	No. of channel	32	64
Central Office Trunk (Lines)	Loop Start	56	112
	DID w/4DIDA	28	56
	2W/4W E&M	14	28
DTI **Note 1**	1.5M	7	10
ISDN	1.5M(PRT)	7	8
	4BRT (card)	5	10

Note 1: *The total number of trunk line and DTI channel shall be 256 or less.*

Least Cost Routing (LCR)

The *UNIVERGE* NEAX 2000 IPS provides extensive Least Cost Routing capabilities. LCR allows the *UNIVERGE* NEAX 2000 IPS to be programmed to route outgoing calls over the most economical facility (WATS, FX, and DDD), based on the area code and office code dialed. In the case of Specialized Common Carriers, associated NXX patterns can be screened out via 6-digit Toll Restriction if the dialed number is not within the designated service area of that SCC.

The *UNIVERGE* NEAX 2000 IPS system provides one of the most flexible route selection capabilities of any system on the market today. While many systems can add and/or delete digits on incoming calls, it is the capability of what can be done on outgoing calls that make the *UNIVERGE* NEAX 2000 IPS such a powerful and cost effective system.

When multiple routes are to be used for an outgoing call, the system is programmed with route order priority (RA). Priority 1 usually being the most cost effective route for the call being made. Up to eight route choices can be assigned for a call. These choices would be put into an Outgoing Route Pattern (OPR). The system allows for up to 64 different OPRs to be constructed.

For true cost effective use of outgoing dialing, it may be desired to change the outgoing route selection during specific times when certain carriers may offer cheaper rates. These OPRs can be assigned using one of eight different Time of Day Patterns (TDPTN). In TDPTN data, you assign what pattern is used at what time of day. You then assign the Time Pattern in the OPR data.

Outgoing calls made using the Least Cost Routing (LCR) can have up to 24 digits added to the number dialed. In OPR data, the system will ask if and how many digits are to be deleted from the number dialed. It will then ask if any digits are to be added, and if so, what are the digits to add in front.

Besides the flexibility of getting a call out of the system, restriction controls make this system one of the best. There are various types of call restrictions that can be made. Some of them are as follows:

Six Digit Least Cost Routing

This allows you to restrict specific area code(s) from going out a specific route, even though the route is included in the OPR constructed. This table lets you choose the TDPTN, the OPR, the RA, and the area code. Then, designate whether the area code is allowed or not.

Route Restriction Class (RSC)

There are 8 different restriction classes that can be created to allow, deny, or toll restrict certain users from dialing certain numbers. An RSC is assigned to the stations. The RSC table is then constructed on a per route basis to determine whether a number dialed from a station in a specific RSC is to be allowed, denied, or toll restricted.

Toll Restriction

The toll code restriction data allows the system administrator to program certain restrictions for Tandem (trunk-to-trunk) and Outgoing calls. Outgoing toll restriction is based on a station RSC. For outgoing, the parameters request the Route Number, Destination Code, RSC, and Toll Data Index (TDI). The TDI can be assigned as; Connection is restricted, Connection is allowed, 3/6 Digit toll restriction, or C.O operator call. When 3/6 Digit toll restriction is used, the system will ask what digit to check the restriction upon (3rd or 6th).

3/6 Digit Toll Restriction

3/6 Digit Toll allows restrictions to be assigned starting with a specific digit place in the dialed number. This command will request the Route Number, RSC, Digit Code (3 or 6 digits), and whether it is allowed or restricted. For example, if I wish to restrict stations with an RSC of "4" from dialing 555 numbers only if a specific route (Route n) gets chosen in the LCR sequence: In the Toll Restriction data, assignment would be to check the 6th digit (9 + 1 + 201 + 5XX + XXXX). In the 3/6 digit toll table assign Route n, RSC4, DC 555, RES: 0 [restricted].

Priority Restriction Class

These provide 8 levels of control over a user's route access eligibility. The system allows a call to reach its destination if the RSC is greater than or equal to the Priority Restriction Class of the outgoing trunk group (route).

Chapter 7 Attendant Answering Position

Attendant Console (SN716)

The Attendant Console (SN716 DESKCON) has an ergonomic design and provides full access to all PBX Console features. The SN716 DESKCON uses the same interface cards as the digital phones to connect to the PBX. The SN716 Desk Console uses a 6-core Modular Jack; inner 1 pair for signal and outer 2 pairs are for power supply to connect to the PBX system.

The SN716 Desk Console operates on a switched-loop basis with a maximum of 6 Attendant loops terminating at each console on the associated Interface card. The Attendant uses these loops for answering, originating, holding, extending, and re-entering calls. When Attendant loop release is used, the number of loops is effectively increased to a maximum of 12 for each console.

SN716 DESKCON General Features

- Character LCD (4 x 40 character)
- LCD designation strips
- Software-controlled LCD loop key
- Full access to PBX features
- Headset connectivity
- Recorder connectivity

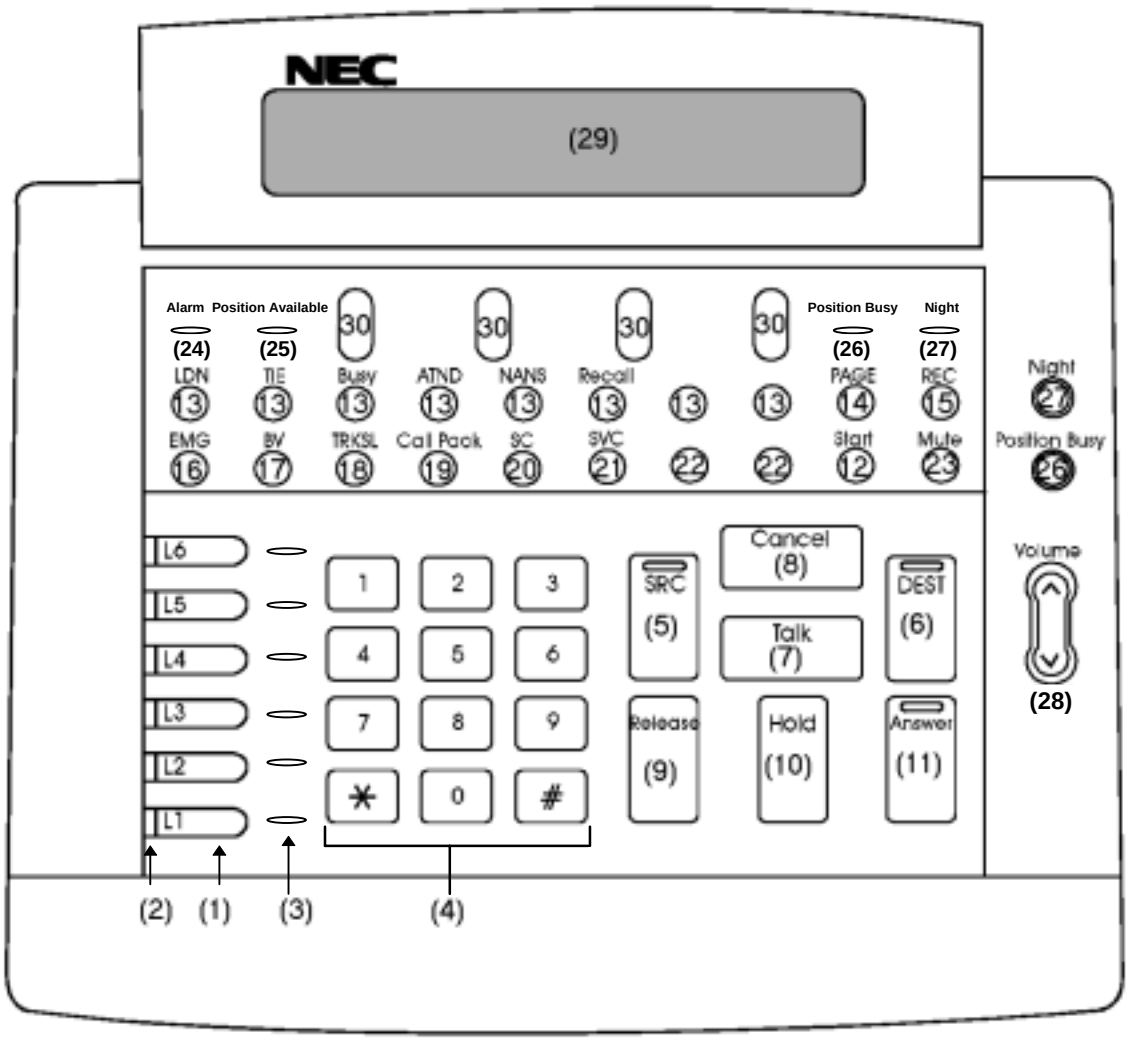
SN716 DESKCON Exclusive Features

While the DESKSON has full access to PBX features the SN716 DESKCON has the following exclusive features.

DESKCON Exclusive Features	
Attendant Assisted Calling	Call Waiting Display
Attendant Camp-on (Full and Semi-automatic)	Common Route Indial
Attendant Called/Calling Name Display	Dialed Number Identification Service (DNIS)
Attendant Called/Calling Number	Incoming Call Identification
Attendant Call Selection	Individual Trunk Access
Attendant Console Lockout-Password	Multi-Function Key
Attendant Do Not Disturb Setup and Cancel	Multiple Console Operation
Attendant Interposition Calling/Transfer	Pushbutton Calling - Attendant Only
Attendant Lamp Check	Serial Call
Attendant Listed Directory Number	Time Display
Attendant Loop Release	Trunk Group Busy Display
Attendant Programming	Unsupervised Trunk-to-Trunk Transfer By Attendant
Attendant Training Jacks	Attendant Delay Announcement
Audible Indication Control	Attendant Lockout
Call Processing Indication	Attendant Overflow
Call Queuing	Attendant Override
Call Splitting	

Note: *For Detail of each feature refer to Chapter 9 Feature Description.*

Face Layout of the SN716 DESKCON



Functions and use of Keys and Lamp Indications

Location Number	Key or Lamp Designation	Full Name	Basic /Option	Key/Lamp	Function
1	L1-L6	Loop	Basic	Key	The attendant answers the call associated to the particular loops. Loop keys are usually used to reenter to held calls, answer automatic recalls.
2	L1A-L6A	Loop Lamp A	Basic	Lamp (one per loop)	Steady green lamp indicates attendant connected to the loop, or called station has answered. Flashing green lamp indicates call waiting to be answered. Steady red lamp indicates call party busy. Flashing red lamp indicates call held at the console.
3	L1B-L6B	Loop Lamp B	Basic	Lamp (one per loop)	Flashing red lamp indicates automatic recall has been activated.
4	Push-button Dial	Push-button Dial	Basic	Key	Allows the attendant to: ▪ Process incoming calls ▪ Originate calls ▪ Activate various service features
5	SRC	Source	Basic	Key & Lamp	Allows the attendant to speak with the calling party. The associated lamp lights when the attendant is connected. The source trunk/station number will be shown in the number display field.
6	DEST	Destination	Basic	Key & Lamp	Allows the attendant to speak with the called party. The associated lamp lights when the attendant is connected. The destination station/trunk number will be shown in the number display field.
7	Talk	Talk	Basic	Key	Allows the attendant to join in a three-way conference with the calling and called parties. When connection is established, both SRC and DEST lamps will light.
8	Cancel	Cancel	Basic	Key	Allows the attendant to: ▪ Disconnect the calling (source) or called party (destination) from the loop. ▪ Disconnect an outgoing trunk or tone seized by the attendant. ▪ Disconnect the station recalling attendant for transfer assistance.
9	Hold	Hold	Basic	Key	Allows the attendant to hold a call at the console and/or to activate it to serial call state.
10	Release	Release	Basic	Key	Allows the attendant to release from an established connection freeing the console for processing of new calls.
11	Answer	Answer	Basic	Key & Lamp	Allows the attendant to answer incoming calls in the order in which they arrive at the console.
12	Start	Start	Basic	Key & Lamp	Allows the attendant to extend an outgoing call to a station. Completion of outgoing pulse will be recognized.

13	Incoming Call Identification: These eight non-locking keys with associated lamps provide attendant access to specific types of incoming calls. A flashing lamp indicates a call waiting to be answered. A steady lamp indicates a call answered. The standard arrangement of these keys is shown on the face layout.				
	LDN	Listed Directory Number	Basic	Key & Lamp	Incoming central office trunk call.
	TIE	Tie Line	Basic	Key & Lamp	Allows the attendant to answer incoming Tie Line calls when the distant station dials access digit to the attendant.
	Busy	Call Forwarding-Busy Line	Basic	Key & Lamp	Allows the attendant to answer incoming calls to specified station when the station is busy.
	ATND	Attendant	Basic	Key & Lamp	Incoming station call.
	NANS	Call Forwarding-Don't Answer	Basic	Key & Lamp	Allows the attendant to answer incoming calls to specified station when the station does not answer within the predetermined time.
	Recall	Recall	Basic	Key & Lamp	Incoming station call for attendant assistance in transferring an established outside call to another station.
	Option	Optional	Option	Key & Lamp	Additional incoming special service calls, such as FX.
	Option	Optional	Option	Key & Lamp	Additional incoming special service calls, such as FX.
14	PAGE	Page	Basic	Key & Lamp	Allows the attendant to connect with pager. (overhead paging)
15	REC	Record	Basic	Key & Lamp	Allows the attendant to connect with recorder.
16	EMG	Emergency	Basic	Key & Lamp	Allows the attendant to answer incoming calls from the station where the station leaves the receiver off.
17	BV	Busy Verification	Basic	Key & Lamp	Allows the attendant to enter into station-to-station connection.
18	TRKSL	Trunk Selection	Basic	Key & Lamp	Allows the attendant to individually select a desired trunk.
19	Call Park	Call Park	Basic	Key & Lamp	Allows the attendant to establish incoming call to Call Park. **Note:** *The attendant can connect this call once again by dialing a specific number and individual number of the console from an ordinary extension telephone.*
20	SC	Serial call Set	Basic	Key & Lamp	Allows the attendant to be automatically recalled when the station user replaces the handset, by depressing the key after extending a central office incoming call to the station user.
21	SVC	Supervisory Call Set	Basic	Key & Lamp	Allows the attendant to supervise a call by depressing the key after extending a central office incoming call to the station user.
22	Option	Optional	Option	Key & Lamp	Allows the attendant access to various optional features provided as required. Each key has an associated lamp.
23	Mute	Mute	Basic	Key & Lamp	Allows the attendant to cut off the voice transmission.
24	Alarm	Alarm	Basic	Key	Steady lamp indicates trouble conditions in the PBX.
25	Position Available	Position Available	Basic	Lamp	When the attendant position (console) is available to process calls, the lamp lights.

26	Position Busy	Position Busy	Basic	Key & Lamp	When the key is pressed, the lamp will light, and the console will become not available. **Note:** *Press the button if operators leave their seats.*
27	Night	Night	Basic	Key & Lamp	Allows the console to place in the night answer mode and lights the associated lamp. Releasing the key restores the console to normal operation.
28	Up Down	Up and Down	Basic	Key	Allows the attendant to adjust volume of the receiver, ringer and contrast of the LCD.
29	LCD	Liquid Crystal Display	Basic	Display	The following information will be displayed: 1st line: The kind of party that connects to the attendant, the number of the waiting calls, the date and time. 2nd line: Tenant number, station class of service and station number belonging to the destination (called) party. 3rd Line: Tenant number, station class of service and station number belonging to the source (calling) party. 4th Line: Optional indication, such as trunk busy.
30	Multi-Function Key	Multi-Function Key	Basic	Display	Keys while idle: Mode−allows access to DAY/NIGHT mode and LKOUT (Console Lock out mode) Prog−allows access for programming DISA, System Speed Dial, Date & Time and Tone Ringer Keys while answering or originating: SPB−Out Pulse Short LPB−Out Pulse Long SHF−Flash Over Trunk Keys while calling a busy station: B.V−Busy Verification Keys while calling a DND station: DDOVR−DND override Keys while accessing Hotel feature: RC−Room Cut off MW−Message Waiting DD−Do not Disturb WU−Wake up call RESET−Reset

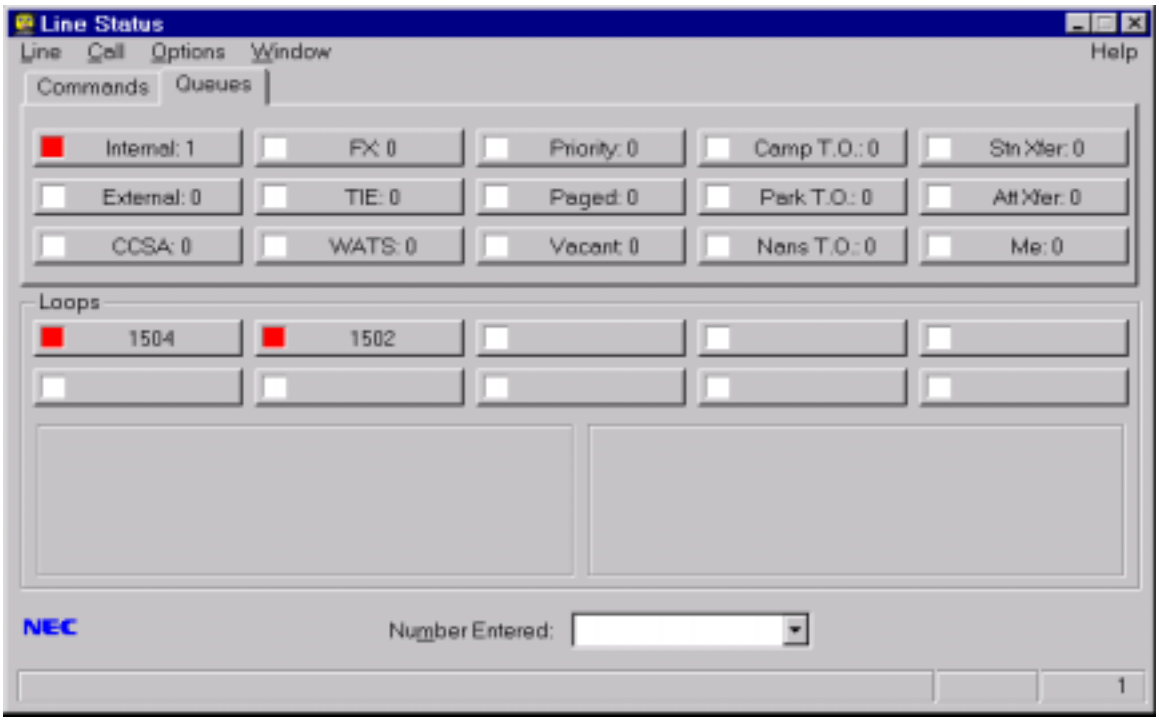
SN716 Specifications

ITEM	Interface and Power Options	Q'ty	Remarks
SN716 DESKCON	PN-8DLCP	1/8 ATT	Interface Card (-27V) Distance: Max. 984.3ft (350m)
	PN-4DLCM	1/4 ATT	Interface Card (-27V) Distance: Max. 984.3ft (350m)
	PN-4DLCQ	1/4 ATT	Interface Card (-27V) Distance: Max. 984.3ft (350m)
	PN-2DLCN	1/2 ATT	Interface Card (-48V) Distance: Max. 984.3ft (350m)
	PN-PW00	1/ ATT	DC/DC Converter (-27V to –48V) Distance with 2/4/8 DLC (350m)
	ACA-U Unit	1/ ATT	24V AC/DC Adapter Distance w/8DLC (350m) Distance w/4DLC or 2DLC (1200m)

- The *UNIVERGE* NEAX 2000 IPS can support up to (8) SN716 Desk Consoles.
- Dimensions: 10 inches (25.4 cm) wide x 9 inches (22.9 cm) deep x 4 inches (10.2 cm) high.

Business Attendant System (BAS)

NEC's Business Attendant System (BAS) is a powerful client/server PC-based tool for full-time and part-time NEAX PBX systems answering positions. The BAS consists of an interactive client side application running on Windows 98 SE, Windows 2000 (SP2) or Windows XP (SP1) operating system that connects to the BAS server via the corporate LAN. The BAS server software is used for PBX connectivity and runs on a Windows 2000 Server (SP2) or Windows 2003 Server. With a dynamic interactive directory and an Open Application Interface (OAI) driven call queue, the operators are able to process a high call volume with minimal effort and time expenditure in today's sophisticated business environment. For example, an operator can answer an incoming call from one of many queues; find out to whom the caller wishes to speak, begin typing that name to select it in the directory, and once selected, hit the <Enter> key or click the entry to transfer the call. Additionally the caller could be parked, put on hold, or sent to voice mail. Several operators can monitor the same queues and share the same directory for call operation. These few examples barely scratch the surface of the capabilities and features available for the system operator positions.

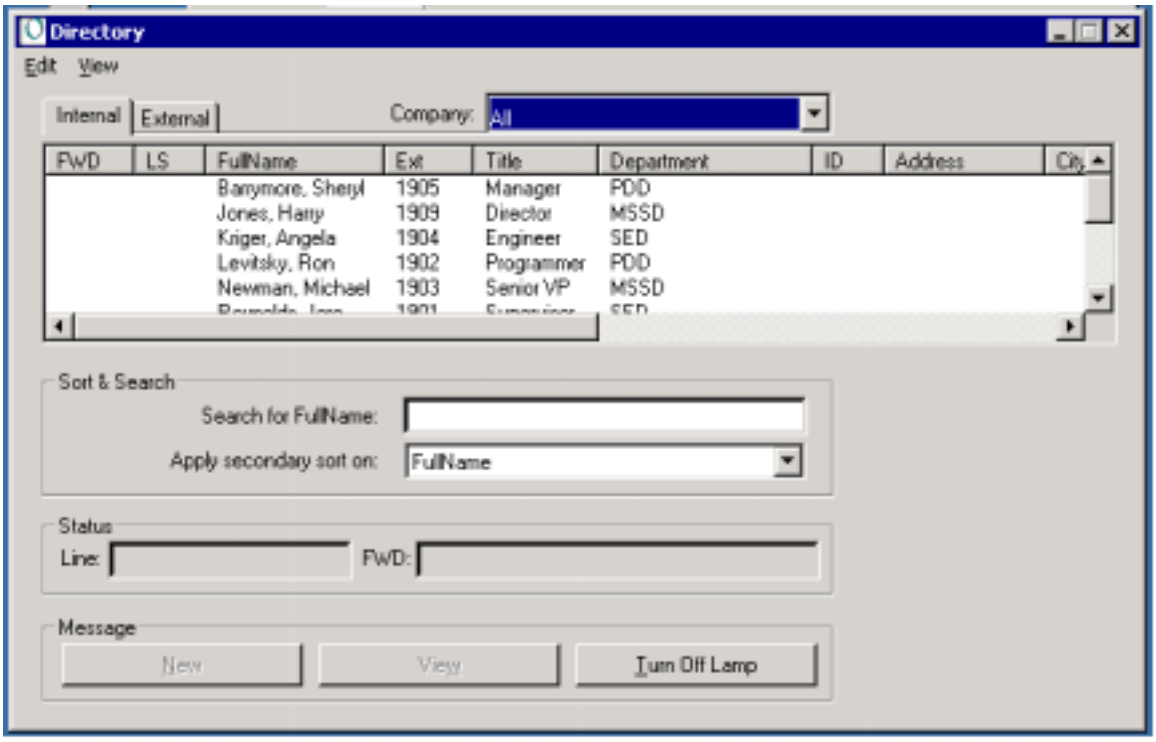


BAS: Benefits

- Centralized company database that can link and replicate with other databases, including AIM*WorX*.
- Sound playback and recordings on the operator telephone.
- Provides operators with the ability to do more than answer the phone, since the client resides on a PC.
- Meet-me paging to enhance operator productivity.
- *Unlimited*, global parked calls.
- Different greetings/displays available for DID calls to the operator, which is especially useful in a multi-tenant environment.
- Single key call processing.
- Directory dialing for accuracy.
- Look ahead status from the directory that saves time, key strokes and enhances decision-making.
- Sort and searchable directory by key fields.
- Configurable screens for *each* operator.
- Time stamped recordings with caller ID and optional note field that can be saved permanently or used for sampling quality.
- Queue types allowing flexibility to answer certain call types first if so desired.
- Configurable information about the caller is displayed upon answering the call allowing personalized service.
- MAPI compliant e-mail integration.
- Ability to set/cancel call forwarding and DND for other phones within the business.
- Delay announcements capabilities to ensure callers are informed and provided other options during busy call times.
- Configurable night station or announcement when operators are not logged in.
- Flexible DID number assigned to each operator that follows them regardless of what PC/phone they use to log in.
- Optional statistics package to track queue and operator statistics (next release).
- Optional integration with Open*WorX* Location Status Information to see an employee's availability, return time, and alternate phone number.
- Optional integration with Open*WorX* Dialer providing Dterm users' access to the same employee and external database .It is searchable by site, department or name.
- Optional integration with Open*WorX* Short Text Messaging to notify wireless users of messages.

Business Attendant System: Directory

Both an internal directory and an external directory exist in the BAS system. The internal directory lists employees within the company, while the external directory stores numbers outside the company. External numbers are usually numbers that are frequently called, or information about people or companies that frequently call the main listed number. When answering a call from one of these external numbers, fields from this entry are displayed to the operator, providing more information than just the 10-digit caller ID. Below is an example of the internal directory screen.



Many of the title bars at the top of the screen are capable of producing a directory sort by clicking the mouse on the title bar. The information displayed is configurable by the administrator. There are several different fields available for display and four of these fields can be user-defined. This window is also expandable for viewing more fields. The “Search” field searches dynamically as the information is typed in the field selected.

The directory is also capable of being partitioned by company. The drop down box at the top of the window displays entries from a particular company, or all companies in the system. This is particularly useful in shared tenant installations, where a single attendant group serves multiple companies.

Business Attendant System: Speed Dials Pad

Another versatile screen is the Speed Dial pad shown below:

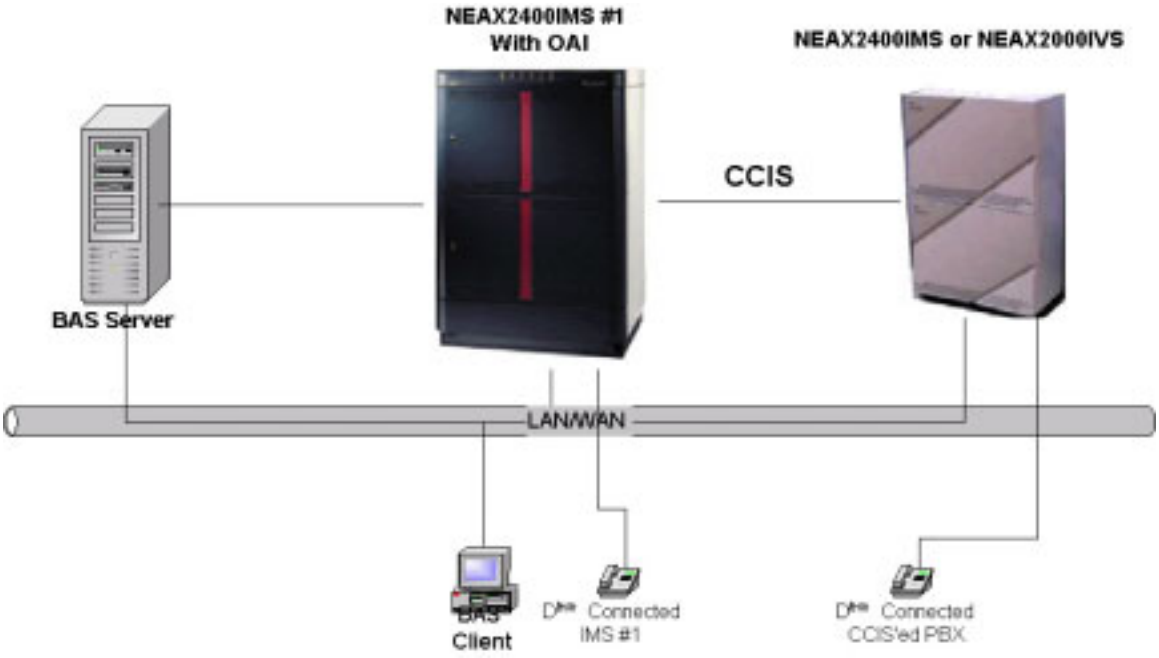
Up to forty (40) Speed Dial fields are available for each attendant. Dragging an entry from the directory onto the speed dial button automatically programs these fields. Another great feature of these buttons is that they act as Busy Lamp Fields (BLF). When that user's phone is off hook, the speed dial button becomes red and when that phone is ringing, the speed dial button blinks red. By right clicking the button, the operator can enable or disable monitoring, as well as change the name, forward the person's Dterm, set DND, or initiate a screened, blind, or announced voice call. This is useful for an operator who also has a set of VIPs for whom they answer calls.

Business Attendant System: Additional Features

As the BAS continues to grow, more features will be added and existing features will be enhanced. Currently, there are several adjunct applications under development for use directly with the BAS directory. Apart from the directory, another powerful feature integrated into the BAS client is the ability to record conversations and playback greetings over the telephone. With the addition of a Dterm Play/Record Module, the client will be able to automatically record voice conversations and instruct the software to save the last specified number of conversations. Conversations are saved as .wav files, which are time; date stamped and rotated in a circular buffer. When it is determined that a call needs to be saved, the operator simply presses the "Save Recordings" button and the selected call is automatically saved. Conversely, the operator can save a set of greetings to be played back immediately every time the phone is answered via the BAS. Greetings playback can be determined based on "internal" or "external" calls, on groups of caller IDs, or even morning or afternoon. A different greeting can be recorded for any of the scenarios above allowing the attendant to present a pleasant and consistent greeting over the telephone.

Business Attendant System: CCIS Network Information

The following is a diagram of a simple CCIS network using the BAS.



The Business Attendant System can run in the above-pictured environment with the following limitations.

- Calls transferred from the BAS Client to a Dterm on the second PBX will not recall to the BAS operator.
- The Dterm in the second PBX cannot be monitored by the BAS system and the monitored speed dials and status checks in the directory will not function for those extensions.
- The system will force supervised transfers to CCIS extensions to ensure callers do not get transferred to a busy remote extension. Operators need to listen for ring back tone or busy tone, and then either completes the call or return back to the source caller as appropriate.
- The message waiting lamp cannot be controlled on remote Dterm's.
- Using multi-node capabilities, the operator can now view station data via the directory or speed dials to extensions in other PBXs. The other PBXs must have OAI and a TCP/IP connection back to the Open*WorX* server. Additional Open*WorX* software is also required per PBX to be added on the server (Short Text Message and Nurse Call do not have multi-node capabilities).

Business Application Environment

Description
Business AE UNIVERGE NEAX 2000
OWX BAS 1 Attendant (12 user Max)
OWX Dialer
OWX Incoming Call Assistant (ICA)
OWX Location Status Information (LSI)
OWX Message Reader
OWX TAPI Link Single PBX
OWX TAPI Link Multi-Switch
OWX Personal Call Assistant (100 user Max)
OWX Group Call Forward Control (GCFC)
OWX Name Display
OWX Short Text Messaging (STM) Email Interface
OWX Short Text Messaging (STM) TAP Interface
OWX Short Text Messaging (STM) Email Upgrade
OWX Short Text Messaging (STM) TAP Upgrade

Hardware Requirements

Description	Qty	Remarks
PZ-M606-A	1	Ether card 1 per system For Peer to Peer connection and MAT/OAI

Chapter 8 System Administration

System Administration

In this system, the Customer Administration Terminal (CAT) or Maintenance Administration Terminal (MAT) is used for programming the system data. The CAT is a digital multi-function telephone (Dterm) which is equipped with function keys, a dial pad and LCD and interfaces with the system via the MP card. The Maintenance Administration Terminal (MAT) is a personal computer that provides an interface to the PBX via the system CPU card. The MAT PC must have the MATWorX program properly installed to communicate with the PBX. MATWorX is required for system software registration and activation.

Password Entry

In a system with password service, a maintenance person is required to enter an authorization level number (Password Level) and appropriate password prior to engaging in programming the system data with the MAT/CAT. A maximum of eight (8) Password Levels can be set up. The number of commands that the maintenance person can access is determined by the Password Level.

Resident System Program

This resident system program generates system data automatically according to the system hardware configuration, thereby providing immediate operation and shorter programming time. When activated, the system scans hardware configuration (such as line/trunk card location) and assigns the system data (such as station numbers, trunk numbers, etc.) according to a predetermined generic program assignment.

Service Conditions

- This service is applicable for equipment installed in PIM0 through PIM3.
- Data for any vacant slot is not assigned.
- Virtual stations are not assigned.
- Some line/trunk cards (PN-AUCA/PN-DK00/PN-CFTA/PN-CFTB/PN-2AMP/PN-4DAT/PN-4RSTF-A/PN-4VCTI/PN-32IPLA) are not assigned, even if mounted.
- An application card is not assigned, even if mounted.
- No tenant assignment is provided. (Tenant 01 is assigned)

Customer Administration Terminal (CAT)

The Customer Administration Terminal (CAT) is a digital multi-function telephone (Dterm) which is equipped with function keys, a dial pad and LCD and interfaces with the system via the MP card. Programming of the system can be done from selected Multiline Terminals with LCD. The designated Multiline Terminals can be placed in program mode, and system data can then be changed. To prevent unauthorized changes, password levels are assigned, providing authorization for access to certain areas of programming and denying access to others.

Service Conditions

1. Programming from a Customer Administration Terminal can only be used when the system is online.
2. All Multiline Terminals with LCD scanned during initialization will be Customer Administration Terminals.
3. The commands CM00 (Office Data All Clear) and CM01 (Office Data Partial Clear) cannot be accessed from the CAT. The CAT cannot delete itself from the system program.
4. Only two Customer Administration Terminals can be in program mode at the same time.
5. The data that can be changed from the CAT can be limited by the Password level assigned. There are eight levels of Passwords that can be assigned in system programming. The relation between Password level and access to available commands is also assigned in system programming.
6. A password can consist of a maximum of any eight digits with the following limitation: The password cannot be CCCCCCCC or FFFFFFFF.
7. Caution should be exercised when assigning Passwords to command authorization levels. If a password is forgotten, access to system programming will be limited and a system initialization with subsequent programming may be required.
8. When the Customer Administration Terminal is offline for programming, it cannot access normal terminal functions.

Maintenance Administration Terminal (MAT)

The Maintenance Administration Terminal (MAT) is a personal computer (PC) that is used for programming and maintenance of the *UNIVERGE* NEAX 2000 IPS system. The MAT can provide a Maintenance Printout, Peg Count information and fault message output. Additionally, the MAT can be used to Remove and Restore to service any station in the system and can read or save system data from disks. The MAT can assign the Key Data for the Attendant Console. The MAT requires an IBM or compatible PC running Microsoft Windows 98, NT 4.0, 2000 or XP and MAT*WorX*.

MATWorX is a Graphical User Interface (GUI) program that provides an efficient method for manipulating the PBX database. This program contains extensive help files, Usage Wizards and Tool Tips, with hyperlinks imbedded in the text. The hyperlinks provide quick access to the appropriate Add-In modules. Add-In modules provide a user-friendly, intuitive method for customizing the PBX database.

MATWorX add-ins makes it easy for you to add or remove PBX features at any time. An add-in is a modular component that lets you program specific features such as Caller ID, Station Assignments, Day/Night Modes, and Line Key Assignments.

Because an add-in is modular, you can add, remove, and upgrade them individually from within MATWorX. An Add-in lets you modify your PBX's features without having to upgrade the MATWorX application itself. MATWorX also gives you a convenient way to launch other commonly used applications, such as Microsoft Word or Excel, directly from its Toolbar.

There are three ways to connect your PC to an NEC PBX:

- Use a modem to establish a dial-up connection.
- Use a serial cable to establish a direct connection.
- Use TCP/IP over your Local Area Network (LAN)

The method you use depends on how you installed and configured the device to which you want to connect. A serial cable direct connection offers better performance than a modem connection, but requires that the PC and device be within 50 feet of each other. A TCP/IP connection offers excellent performance and flexibility but requires a network connection to both your PC and the device.

PBX Configuration Wizard

The PBX Configuration Wizard is a custom tool in MAT*WorX* that enables you to establish the proper communication settings between your computer and the *UNIVERGE* NEAX 2000 IPS. The Wizard asks you simple questions and then uses the information to automatically configure the connection for the PC and the PBX.

Service Conditions

1. Connection through modems is available, providing remote maintenance capabilities.
2. MAT*WorX* can be connected to the system either directly or remotely. Direct connection is through the RS connector on the MP card. Remote connection is available via either an internal modem on the MP card or an external modem for high speed. Remote connection via the internal modem is through the COT card. Connection between the modem and the COT is accomplished through internal switching of the TDSW. Remote connection via an external modem is through the MP card.
3. The following functions can be performed from MAT*WorX*:
   - System, station, and trunk data entry, change, and copy.
   - Loading, saving, and verification of system data to and from a disk.
   - ROM check readout of generic program.
   - Display of fault/fault cleared messages.
   - On-site or remote access to the system.
   - Printout of system data (when printer is connected to PC).
   - Display and setting of system clock/calendar.
   - Numbering Plan
   - Least Cost Routing (LCR)
   - System initialize
   - UCD/Station Hunting/Call Pickup – Group
4. The PC used with MAT*WorX* must have an RS-232C port, and cannot be located more than 50 feet (15m) from the system when connected on premises.
5. When stations or trunks are expanded, moved, or changed, office data for a Multiline Terminal key/station/ trunk can be copied and multiple assignments of related office data is possible.

MACH Script Editor

This is a powerful timesaving tool that enables you to create a list of *UNIVERGE* NEAX 2000 IPS commands that perform tasks in the PBX. This list is referred to as a script, which can be saved and run at anytime. You can also use the MACH Script Editor to perform many other operations.

Traffic Management

The *UNIVERGE* NEAX 2000 IPS provides traffic management reports to be used for overall analysis of system performance. MAT*WorX* is used to request and display the type of report, sample measurement time period, and time increments of reports. Type of Traffic Measurement Reports:

- Number of successful attempts at outgoing access, based on trunk route.
- Number of times all trunks were busy, based on trunk route.
- Number of incoming calls, based on trunk route.
- Number of incoming calls connected to busy tone and then trunk abandoned.
- Quantity of incoming calls with no answer and trunk abandoned, based on trunk route.
- Number of times a push button register was connected, on a system basis.
- Number of times all push button registers were busy, on a system basis.
- Number of outgoing connections using modem trunks, based on modem trunk group.
- Number of incoming connections using modem trunks, based on modem trunk group.
- Number of times all modem trunks were busy, based on modem trunk group.
- Number of times the Conference circuits were used, on a system basis.
- Number of times Conference circuits were all busy, on a system basis.
- Number of times an incoming call was Call Forwarded-No Answer to the Attendant or another station (on DID, Tie or DIT lines), on a system basis.
- Number of Tandem Connections, on a system basis.
- Number of times a push button register was connected to a trunk, based on trunk route.
- Number of Attendant calls including recalls, on a system basis.
- Number of station-to-station calls, on a system basis.
- Number of times senders were all busy, on a system basis.
- Number of ring generator capacity overflows, on a system basis.
- Number of DTE to DTE connections, on a system basis. (UCD Peg Count)
- Number of answered calls by UCD group.
- Number of incoming calls by UCD group.
- Number of waiting calls for a pre-determined time into queuing mode on the UCD group.
- Number of incoming calls to all busy stations in the UCD group.
- Number of answered calls in the UCD group.

MATWorX PC Requirements

MAT*WorX* requires an IBM or compatible PC running Microsoft Windows®ME, 2000 or XP.

Minimum Requirements	Recommended
Pentium III 350 MHz processor	Pentium IV 1 GHz processor
128 MB RAM	256 MB RAM
Available hard-disk space before installation: 200 MB - Full Installation MATWorX uses about 100 MB	Available hard-disk space before installation: 200 MB - Minimum installation MATWorX uses about 100 MB
SVGA monitor, 800 x 600 resolution	15” SVGA monitor, 1024 x 768 resolution
3.5” diskette drive, CD-ROM drive, Mouse	
Valid NEAX hardware connection (direct serial, modem, or TCP/IP)	

System Diagnostics

When a fault occurs in the system, an audible and visual indication will be given at the following units:

- External alarm indicating unit
- Fault messages reported at MAT*WorX* for remote reporting
- Alarm lamps in front of each package mounted in the frame

Self Diagnostic/System Messages

The *UNIVERGE* NEAX 2000 IPS provides a sophisticated array of self-diagnostic routines that are continually and automatically being performed. Various system messages are printed when a fault occurs in a central processor system, switch network processors. Many other miscellaneous system messages and change of key status messages are also printed.

Remote Maintenance

This feature allows station and trunk changes or reassignments to be performed without a site visit by service personnel, and can be used to retrieve fault codes prior to visiting a site. One Remote Maintenance center can service an unlimited amount of systems, thus reducing the amount of personnel to maintain each site, travel costs and customer billing for each site.

Service Conditions

1. The following additional equipment is required for this feature:
   - A modem at the maintenance center and one at each remote site. (When the internal modem of the Main Processor (MP) is used, no modem at each remote site is required)
   - A cable for connection between the MP and the on-site modem. (When the internal modem of the MP is used, the above cable is not required)
2. The internal modem of the MP is compatible with the following specifications:

Modem Specification	
▪ ITU-T V.22 1200 bps	▪ ITU-T V.32 4800/9600 bps
▪ ITU-T V.22 bis 2400 bps	▪ ITU-T V.34 19.2 k/33.6 kbps
▪ Bell 212A 1200 bps	

3. Any one of the following connections are also required for access to the modem:

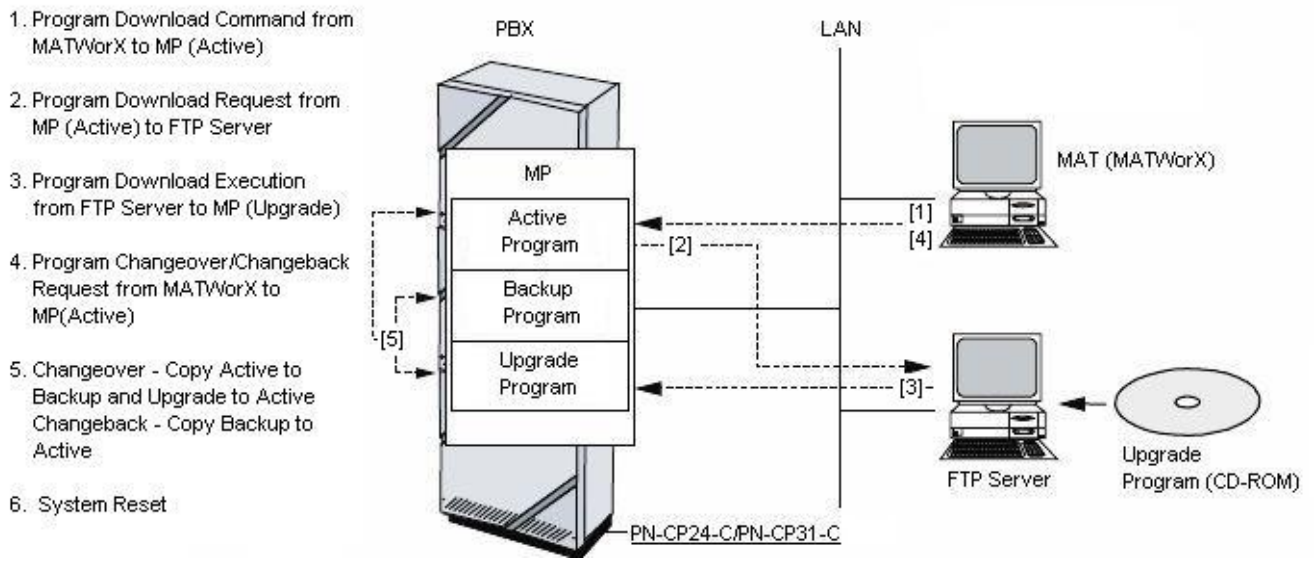
Required Connection	
▪ A dedicated line	▪ Attendant controlled transfer
▪ Direct Inward System Access (DISA)	▪ Direct Inward Termination

4. The following operations can be executed from the Remote Maintenance location:
   - Retrieval of fault data
   - Retrieval of Peg Count information
   - Deletion or addition of system data (line, trunk, etc.) using a preprogrammed security password
   - Data assignment by device number (stations, trunks, and Attendant Console)
   - Copying of station data from one station to other stations (when adding sequential stations in groups)
   - Release / Reconnection of backup batteries
   - Display of station line status

MP Program Download

The *UNIVERGE* NEAX 2000 IPS provides Online MP Program Download via IP network using the CP24C and CP31C. This feature allows an MP upgrade program to be downloaded to the *UNIVERGE* NEAX 2000 IPS MP card with the PBX on-line and during the download process all features and functions are available. The MP Program Download feature is available to a stand alone system or a Remote system in a Remote PIM network.

The IPS downloads the MP upgrade program from FTP server using MATWorX. MATWorX and the FTP server can be installed and running on the same PC. A single FTP server or multiple FTP servers can be used. Immediate or scheduled changeover to the upgrade program is available. It is also possible to change back to the previous program that was in use before the changeover (changeback).



MP Download Process

Service Conditions

General Service Conditions

- The MP program download can be executed to the PN-CP24-C/PN-CP31-C (MP card) when a PZ-M606-A is mounted.
- For the Retrofit system and Backup CPU system remote download of MP program is not available.
- The call processing is stopped while the system is initialized by the program changeover execution.
- Remote download of AP program is not available.
- While the PBX is off-line, you cannot execute the program downloading, or program changeover/ changeback/ program version matching (immediate). However, it is possible to read various information and set the schedule (date/time) for program changeover/ changeback/ program version matching even while the PBX is off-line.
- When the PN-32IPLA is mounted in the system, the MP program download (FTP) is not available. When the PN-32IPLA-A or PN-8IPLA is mounted, it is available in the system.

Program Download Service Conditions

- Do not reset the PBX and the MATWorX during MP program downloading.
- Do not pull out the LAN cable during MP program downloading.
- FTP (File Transfer Protocol) is used as the protocol for the remote download of MP program.
- Only the file specified in "File Name" of CHECKSUM.TXT can be downloaded to the PBX.
- If you do not set the directory name to save the program files and CHECKSUM.TXT by MATWorX, they are saved to the FTP directory (default).
- If you do not set the password and user ID of the FTP server by MATWorX, login to the FTP server with anonymous.
- If required FTP server information is not set when executing the MP program download, "DATA NOT SET" is displayed.
- For the TCP port, the specified port is used for file transfer (control), and the port in front of it is used for the file transfer. For example, when the TCP port is set to 3000, Port No. 3000 is used for the file transfer (control), and Port No. 2999 is used for the file transfer (Port 21-File transfer (control), Port 20-File transfer: default).
- If you read the issue number of the standby side (outdated side) and change the settings of FTP information during MP program downloading, "WAIT, BUSY NOW" is displayed.
- When using a single FTP server to update multiple sites in remote PIM network schedule the file transfer at least 7 minutes or more apart so that bandwidth and the number of connections to the FTP server are not exceeded.

Program Changeover Service Conditions

- The system is initialized automatically after executing the program changeover any data that is not backed up will lost. Be sure to execute the office data backup before executing the program changeover.
- If an error is detected during system initialization after executing the program changeover, the PBX starts automatically with the previous program.
- If the program download is executed while the program version matching is being executed, the program version matching is interrupted, and program download is executed.
- If the program changeover (immediate) is executed while the program version matching is being executed, the program is changed after the program version matching is completed.
- The program changeover cannot be executed if the program downloading or program version matching is interrupted or fails. If the program changeover (immediate) is set, the setting is invalid.
- The program changeover for which the schedule (date/time) has been set is executed only when the PBX is on-line.
- When immediate program changeover is executed after a schedule (date/time) for program changeover execution is set, the schedule is invalid.

Required Hardware

Equipment		Comments
CP24C/CP31C		Stand Alone or Remote PIM network
MATWorX IPS		Version 11.0.0
PC for MATWorX	**Note**	See MATWorX PC Requirements above
PC/Work station for FTP	**Note**	Windows 2000 Server/Windows 2000 Professional/Windows XP Professional (Internet Information Server (IIS)supplied with Windows 2000/XPProfessional can be used)

Note: *MATWorX and FTP server program can be used on the same PC.*

MA4000 Management System

MA4000 Management System is a web based, powerful voice server management configuration suite. MA4000 offers centralized management for the *UNIVERGE* NEAX 2000 IPS Voice Server as well as simple and powerful tools for managing moves, adds and changes. MA4000 also offers flexibility and security and allows the end user to manage their voice server in the same ways they manage their networks.

- Web Browser Operation
- Email Alarm Notification
- Alarm Client Notification
- LDAP Auto Provisioning Service
- LDAP Authentication
- Windows Authentication
- IT Friendly Interface
- Flexible Access Levels (Roles)
- Application Program Interface (API & SDK)
- Security
  - HTTPS supported
  - Audit Trail Logging
- Customizable Portals
- Command Line Interface
- Integration to OpenWorX
- Centralized Authentication Service (NEC CAS)
- System Health Monitoring
- Range programming that is schedulable

The MA4000 IPS Assistant gives the power of station management directly to the end-user. Limited by access rights, an administrator can allow his users to manage their own phones. Supported features include Button Programming, Call Forwarding and Speed Dial Programming.

Chapter 9 Feature Descriptions

Business/Hotel/Data Feature List

Account Code	Attendant Delay Announcement
Add-On Module	Attendant Lockout
Alarm Indications	Attendant Overflow
Alphanumeric Display	Attendant Override
Analog Port Adapter	Authorization Code
Announcement Service	Automated Attendant
Answer Key	Automatic Call Distribution (ACD)
Attendant Assisted Calling	Busy In/Busy Out - ACD
Attendant Camp-On (Full and Semi-automatic)	Call Waiting Indication - ACD
Attendant Console	Delay Announcement - ACD
Attendant Called/Calling Name Display	Hunt Past No Answer - ACD
Attendant Called/Calling Number	Immediate Overflow - ACD
Attendant Call Selection	Priority Queuing - ACD
Attendant Console Lockout-Password	Queue Size Control - ACD
Attendant Do Not Disturb Setup and Cancel	Silent Monitor - ACD
Attendant Interposition Calling/Transfer	ACD w/Management Information System (MIS)
Attendant Lamp Check	Automatic Camp-On
Attendant Listed Directory Number	Automatic Change to Daylight Saving Time
Attendant Loop Release	Automatic Number Identification (ANI)
Attendant Programming	Automatic Recall
Attendant Training Jacks	Automatic Wake-Up
Audible Indication Control	Background Music
Call Processing Indication	Back Up CPU
Call Queuing	Bandwidth Control
Call Splitting	Boss/Secretary Calling
Call Waiting Display	Broker's Call
Common Route Indial	Call Back
Dialed Number Identification Service (DNIS)	Call Forwarding
Incoming Call Identification	Attendant Call Forwarding Setup and Cancel
Individual Trunk Access	Call Forwarding - All Calls
Multi-Function Key	Call Forwarding - Busy Line
Multiple Console Operation	Call Forwarding - No Answer
Pushbutton Calling - Attendant Only	Call Forwarding - Destination
Serial Call	Multiple Call Forwarding - All Calls
Time Display	Multiple Call Forwarding - Busy Line
Trunk Group Busy Display	Multiple Call Forwarding - No Answer
Unsupervised Trunk-to-Trunk Transfer By Attendant	

Business/Hotel/Data Feature List (Cont.)

Split Call Forwarding - All Calls	Direct Data Entry
Split Call Forwarding - Busy Line	Direct Digital Interface
Split Call Forwarding - No Answer	Direct Inward Dialing (DID)
Call Forwarding - Logout (DtermIP)	DID Call Waiting
Call Forwarding - Override	DID Digit Conversion
Group Diversion	DID Name Display
Call Park	Direct Inward System Access (DISA)
Call Park - System	Call Forwarding Set by DISA
Call Park - Tenant	Direct Inward Termination (DIT)
Call Pickup	Direct Outward Dialing (DOD)
Call Pickup - Direct	Direct Station Selection/Busy Lamp Field (DSS/BLF) Console
Call Pickup - Group	Busy Out Status Console
Call Pickup - Designated Group	Do Not Disturb Console
Call Redirect	Message Waiting Console
Call Transfer	Room Cutoff Console
Call Transfer - All Calls	Wake Up No Answer Console
Call Transfer - Attendant	Distinctive Ringing
Caller ID Class	Do Not Disturb
Caller ID Display	Do Not Disturb - Group
Caller ID - Station	Do Not Disturb - Hotel/Motel
Camp-On	Do Not Disturb - System
Centrex Compatibility	Dterm Assistant
Check In/Check Out	DtermIP
CID Call Back	Elapsed Call Timer
CID Call Routing	Enhanced 911
Class of Service	Executive Calling
Code Restriction	Executive Override
Conference (Three/Four Party)	External Paging with Meet-Me
Conference (Six/Ten Party)	Fax Arrival Indicator
Conference (32 Party)	FAX over IP
Group Call	Feature Activation from Secondary Extension
Meet-Me Conference	Flexible Line Key Assignment
Consecutive Speed Dialing	Flexible Numbering Plan
Consultation Hold	Flexible Ringing Assignment
Customer Administration Terminal (CAT)	Forced Account Code
Data Line Security	Group Call
Delayed Ringing	Automatic Conference (6/10-Party)
Diagnostics	2 Way Calling
Dial by Name	
Dial Conversion	

Business/Hotel/Data Feature List (Cont.)

Group Call by Pilot Number Dialing	Mobility Access
Group Listening	Modem over IP
Hands free Answerback	MP Program Download (FTP)
Hands free Dialing and Monitoring	Multiple Language Display
Hold	Multiple Terminal Attendant Position
Call Hold	Music On Hold
Dual Hold	Night Service
Exclusive Hold	Attendant Night Transfer
Non-exclusive Hold	Call Rerouting
Hotel/Motel Attendant Console	Day/Night Mode Change by Attendant Console
Hotel/Motel Front Desk Instrument	Day/Night Mode Change by Station Dialing
Hotline - Inside/Outside	Day/Night Mode Change by System Clock
House Phone	Night Connection - Fixed
Individual Attendant Access	Night Connection - Flexible
Intercept Announcement	Trunk Answer Any Station (TAS)
Intercom	Overflow for TAS Queue
Manual Intercom	Queue Limit for TAS
Automatic Intercom	No CID Call Routing
Dial Intercom	Off-Hook Alarm
Internal Tone/Voice Signaling	Off-Premises Extensions
Internal Zone Paging with Meet-Me	Open Application Interface (OAI)
IP Enabled Dterm	Pad Lock
Last Number Redial	Periodic Time Indication Tone
Least Cost Routing - 3/6 Digit	Pooled Line Access
Line Lockout	Power Failure Transfer
Line Preselection	Priority Call
Maid Status	Privacy
Maintenance Administration Terminal (MAT)	Direct Privacy Release
Message Center Interface (MCI)	Manual Privacy Release
Message Registration	Private Lines
Message Reminder	Property Management System Interface
Message Waiting	Proprietary Multiline Terminal
Miscellaneous Trunk Access	Automatic Idle Return
CCSA Access	Called Station Status Display
Code Calling Equipment Access	Calling Name and Number Display
Dictation Equipment Access	Group Feature Key
Foreign Exchange (FX) Access	Dynamic Dial Pad
Radio Paging Equipment Access	Hands free Unit
Wide Area Telephone Service (WATS) Access	I-Hold/I-Use Indication

Business/Hotel/Data Feature List (Cont.)

Microphone Control	Station Speed Dialing
Multiple Line Operation	Step Call
Mute Key	Supervisory Control of Peripheral Equipment
My Line Number Display	System Clock Setup by Station Dialing
Preset Dialing	System Speed Dialing
Prime Line Pickup	Tenant Service
Recall Key	Terminal Login via NAT
Relay Control Function Key	Tie Lines
Ring Frequency Control	Tie Line Tandem Switching
Ringing Line Pickup	Timed Forced Release
Soft Keys	Timed Queue
Volume Control	Timed Reminder
Remote Hold	Trunk - Direct Appearances
Remote PIM over IP	Trunk Queuing - Outgoing
Reserve Power	Trunk-to-Trunk Connection
Resident System Program	Uniform Call Distribution (UCD)
Return Message Schedule Display	Busy In/Busy Out - UCD
Room Cutoff	Call Waiting Indication - UCD
Room Status	Delay Announcement - UCD
Route Advance	Hunt Past No Answer - UCD
Save and Repeat	Immediate Overflow - UCD
Security Alarm	Priority Queuing - UCD
Set Relocation	Queue Size Control - UCD
Short Message Service (SMS)	Silent Monitor - UCD
Single Digit Dialing	Uniform Numbering Plan (UNP) - Voice and Data
Single Digit Feature Access Code	Variable Timing Parameters
SIP Trunk Interface	Voice Guide
SNMP	Voice Mail Integration
Software Line Appearance (Virtual Extensions)	Voice Mail Private Password
Stack Dial	Voice Mail Transfer
Station Hunting	VoIP
Station Hunting - Circular	VoIP Log Collection
Station Hunting - Terminal	Whisper Page
Station Hunting - Secretarial	
Station Message Detail Recording (SMDR)	

Business/Hotel/Data Feature Descriptions

Account Code
This feature, when used with Station Message Detail Recording (SMDR), allows station users and Attendants to enter a cost accounting or client billing code (up to 16 digits) into the system.

Add-On Module
This feature allows the Add-On Module to be combined with a legacy Multiline Terminal when there are insufficient line or trunk keys provided at the Multiline Terminal. When the DCU-60-1 unit keys are programmed as line/trunk keys, the additional 25 lines/trunks and the existing lines/trunks set for the Multiline Terminal can be accessed directly (maximum of 49 lines/trunks). The station speed dialing function can be assigned for all keys on the DCU-60-1 unit. Also, one of the last 3 keys can be used as a Day/Night change key.

Alarm Indications
Faults are indicated by the Major/Minor (MJ/MN) lamps located on the AC/DC Power Supply and, optionally, an external alarm display unit.

Alphanumeric Display
The DTR-16DC-1 Multiline Terminals are each equipped with a 3-line, 24-character Liquid Crystal Display (LCD). These displays are used to provide alphanumeric information including clock/calendar and call processing information.
Station Application All Multiline Terminals with an LCD display.

Analog Port Adapter
This feature allows an APR-U/AP(R)-R or APA-U/AP(A)-R unit combined with a legacy Multiline Terminal to connect to an analog terminal such as an analog telephone, Modem, and PC with built-in Modem. There are two communication modes for the terminal connected via the Analog Port Adapter as shown below:

1. Single Port Mode
A Multiline Terminal and an analog terminal share the same port. In this mode, the Multiline Terminal and the analog terminal cannot be used simultaneously.

2. Dual Port Mode
A Multiline Terminal and an analog terminal use different ports. In this mode, the Multiline Terminal and the analog terminal can be used simultaneously.

Announcement Service
This feature allows station users to record messages on Digital Announcement Trunk (DAT) cards. When a station user dials the feature access code for this feature, the user receives the corresponding message from the system.

Answer Key
An Answer Key is provided on all Multiline Terminals. The Answer Key can be used to answer incoming calls on outside lines, and primary or secondary extensions. When the Answer Key is used to answer an incoming call with a call in progress, the first party is placed on hold and the second party is connected. If the Answer Key is depressed while in a three-party call, the user can alternate between each party and a Broker's Call is established.

Attendant Assisted Calling This feature allows a station user to ask an Attendant for assistance in originating a call. Three methods are available: non-delay, delay, and passing dial tone. **Attendant Camp-on (Full and Semi-automatic)** The Full Automatic Camp-on feature permits the Attendant to hold an incoming call in a special mode when the desired station for the transfer is busy. The Attendant sends a Camp-On tone to the busy station. When that station becomes idle, it is automatically alerted and connected to the waiting party. The Semi-automatic Camp-on feature works similarly to full automatic except when the station becomes idle the call on Camp-on recalls to the Attendant. Once the Attendant answers the Camp-on recall the station is then called and the call is connected. **Attendant Console (SN716 DESKCON)** The Attendant Console (SN716 DESKCON) operates on a switched-loop basis with a maximum of 6 Attendant loops terminating at each console on the associated Interface card. The Attendant uses these loops for answering, originating, holding, extending, and reentering calls. When Attendant loop release is used, the number of loops is effectively increased to a maximum of 12 for each console. **Attendant Called/Calling Name Display** This feature provides a display of the calling/called party's name on the Attendant Console LCD for Attendant Called/Calling Name Display. On attendant-to-station calls, the LCD display the name assigned to the primary extension of the station. On attendant-to-trunk calls, the LCD displays the name assigned to the trunk route of the trunk.	**Attendant Called/Calling Number** This feature provides a display of the station number and station name on the Attendant Console during an Attendant-to-station connection. During an Attendant-to-trunk connection, the same display shows the trunk route designation and a trunk identification code (4 digits). **Attendant Call Selection** This feature allows assignment of keys on the Attendant Console to particular types of trunk routes (such as WATS or FX) and particular types of service calls (such as Attendant recalls, intercept calls, etc.). LEDs indicate the type of incoming call and pressing the associated key allows the Attendant to answer the calls in any order. **Attendant Console Lockout-Password** This feature allows the Attendant Console to be set into a lockout mode. This diables the console from originating or receiving calls and setting or resetting service features. To return the Console to its manual operating condition a password is required. **Attendant Do Not Disturb Setup And Cancel** The Attendant has the ability to enter and remove individual stations from Do Not Disturb (DND). Additionally, the Attendant can set one preassigned group of stations into, or out of, Do Not Disturb. **Attendant Interposition Calling/Transfer** This feature allows any Attendant to directly converse with another Attendant and also allows Attendants to transfer calls from their console to another Attendant's console in systems where Multiple Console Operation has been provided.

Attendant Lamp Check

This function is used to check the status of keys, lamps, and LCDs mounted on the Attendant Console to verify that various operations of the Attendant Console are functioning normally. The check is done by a preset procedure.

Attendant Listed Directory Number

This feature provides a display of the Listed Directory Number on the Attendant Console when the operator has answered a Listed Directory Number call.

Attendant Loop Release

This feature allows an Attendant Console loop to become available for a second call as soon as the Attendant has directed the first call to a station, even if that station does not answer.

Attendant Programming

This function is allowed only for the Attendant Console (SN716) and is used to execute DISA code set up, speed dial programming, and system clock set up operations.

Attendant Training Jacks

The Attendant Console can be equipped with two headset/handset jacks using an optional adapter. Two jacks are equipped on the adapter and can be used for training new operators.

Audible Indication Control

This feature allows the Attendant to adjust the volume of audible indications received at the Attendant Console.

Call Processing Indication

This feature provides visual indications of all calls being processed or awaiting processing at the Attendant Console.

Call Queuing

This feature provides the Attendant the ability to handle a series of exchange network calls in the order of their arrival, (first in, first out) thereby eliminating unnecessary delays.

Call Splitting

This feature allows the Attendant to confer privately with one party on an Attendant handled connection without the other party overhearing.

Call Waiting Display

This feature provides a visual indication to the Attendant when one or more calls are waiting to be answered.

Common Route Indial

This feature allows assignment of incoming DID calls to different Attendant Call Selection keys based on the last 4 digits dialed into the system. Up to eight individual Listed Directory Numbers can be assigned in system programming. When an incoming call to any of these trunks is received, an Attendant Call Selection key will flash and the LCD display will indicate the Listed Directory Number associated with that trunk route.

Dialed Number Identification Service (DNIS)

This feature provides a display of the company name on the Attendant Console when the Attendant has answered a Listed Directory Number or a Tie Line call.

Incoming Call Identification

Incoming calls are identified by various means. Refer to Attendant Called/Calling Number, Attendant Call Selection, Attendant Source Key, Attendant Listed Directory Number and Common Route Indial Features and Specifications.

Individual Trunk Access

The Attendant Console is provided with the ability to access each individual trunk by dialing an associated identification code. This allows detection of faulty trunks during regular testing or after complaints. The Customer Administration Terminal (CAT) or Maintenance Administration Terminal (MAT) has the capability to then busy out the trunk until repair is made.

Multi-Function Key This feature allows the top row of keys on the Attendant Console to perform and display multiple functions in accordance with the status of call processing. **Multiple Console Operation** This feature allows more than one Attendant Console to operate within the same system. **Pushbutton Calling - Attendant Only** This feature permits an operator to place all calls over Dual-Tone, Multi-Frequency (DTMF) lines from the pushbutton keypad on the Attendant Console. **Serial Call** This feature is activated by the Attendant when an incoming calling party wishes to speak with more than one internal party. When the internal station subsequently disconnects from the Central Office line call, the Central Office party automatically rings back to the same Attendant. **Time Display** This feature provides a digital time display on the Attendant Console LCD. **Trunk Group Busy Display** A visual indication is supplied to the Attendant when all trunks in a particular trunk group are busy. **Unsupervised Trunk-to-Trunk Transfer By Attendant** This feature allows an Attendant to transfer an incoming or outgoing call on one trunk to an outgoing trunk and exit the connection before the called party answers.	**Attendant Delay Announcement** This feature provides an announcement, via a Digital Announcement Trunk Card, to external calls that are not answered by the attendant within a predetermined time. **Attendant Lockout** This feature denies an Attendant the ability to reenter an established trunk or station connection without being recalled by that station after the call is put in consultation hold. **Attendant Overflow** When an incoming call, which has terminated from a trunk to the Attendant Console, remains unanswered after a predetermined time period, this feature provides a change to Night Service for that particular trunk. **Attendant Override** This feature permits an Attendant to enter a busy connection (station or trunk) using the Attendant Console. When this feature is activated, a warning tone is sent to the connected parties after which, they are connected with the Attendant in a three-way bridge. **Authorization Code** An Authorization Code is a numerical code which will temporarily change a station's Class of Service to a Class of Service assigned to that Authorization Code. This new Class of Service allows access to trunks, dialing patterns, and/or features that would otherwise be restricted. **Automated Attendant** This feature allows the system to answer incoming trunk calls. The system will supply a message and/or dial tone to the caller. The caller can then dial the desired extension number and be directed to that station.

Automatic Call Distribution (ACD)

The Automatic Call Distribution (ACD) feature permits incoming calls to terminate to a prearranged group of stations. Calls are distributed in the order of arrival to idle terminals within the group, based on which terminal has been idle the longest period of time. Stations may log on/log off from the ACD group. Supervisor stations may monitor conversations of agents.

Busy In/Busy Out - ACD

This feature allows an agent in an ACD group to log their station into or out of the group. This allows the system to control whether a call directed to the pilot number of the ACD group goes to that station or not. This prevents incoming calls from being directed to stations at which no agent is available.

Call Waiting Indication - ACD

This feature provides a visual indication when an incoming call to an ACD group is placed in queue, due to an “all agents busy” condition. On external relay controlled indicator or an LED on a Multiline Terminal can be used to provide Call Waiting Indication.

Delay Announcement - ACD

This feature allows the system to provide a recorded announcement to an incoming caller placed in queue to an ACD group. A single announcement, or two separate announcements, can be provided.

Hunt Past No Answer - ACD

This feature allows calls targeted at an ACD group to hunt past an agent’s station, after a no answer condition, if the agent forgets to log off of the group and the agent is unable (or not available) to answer the call.

Immediate Overflow - ACD

This feature allows a call directed to an ACD group to immediately overflow to another ACD group, upon encountering an “all agents busy” condition.

Priority Queuing - ACD

This feature allows the system to prioritize incoming calls by trunk route and on a per station basis, when the call enters an ACD queue. When a call is considered as a priority, it is placed at the beginning of the queue.

Queue Size Control - ACD

On incoming DID/Tie line calls, the system can be assigned a threshold which limits the number of calls in queue. When the queue size threshold is exceeded, incoming callers are connected to busy tone.

Silent Monitor - ACD

This feature provides the ACD group supervisor with the ability to monitor a call to an ACD agent. The silent monitor function gives no indication (as an option) to either the agent or the calling party.

Automatic Call Distribution with Management Information System (MIS)

The Automatic Call Distribution (ACD) with Management Information System (MIS) feature provides a management information system to be used in conjunction with the built-in ACD features of the system. The MIS incorporates a supervisor's terminal for real-time monitoring of agent activity, amber and red alarms, and hard-copy summary reports.

Automatic Camp-on

An incoming Direct Inward Termination (DIT) call which has been terminated to a busy station can be Camped-On automatically. When the busy station becomes idle, the station is automatically called and connected to the camped on incoming trunk call.

Automatic Change to Daylight Saving Time
This feature allows the 2000 IPS system clock to automatically change from standard time to daylight saving time, and vice versa. Schedule to change to/from daylight saving time is programmed by system data programming.

Automatic Number Identification (ANI)
This feature receives the calling subscriber's number automatically sent from T1 network using MF signaling and displays the calling number on the LCD of a Multiline Terminal and an Attendant Console.

Automatic Recall
This feature works as a timed reminder. When a call remains on Hold, Camp-On or ringing unanswered for a fixed interval after being transferred, the station that initiated the hold, transfer, or Camp-On is automatically alerted.

Automatic Wake-up
This feature allows the system to be programmed to automatically call guest rooms or administration stations at specified times. Upon answering, the guest is connected to a recorded announcement or music source. A printout of unanswered or blocked Automatic Wake-Up attempts for each guest room is provided using the Hotel/Motel printer.

Background Music
Background Music can be provided on a dial-up basis over legacy Multiline Terminal speakers. Incoming voice announcements, ringing and recalls override Background Music. Up to 10 music programs can be offered.

Back Up CPU
UNIVERGE NEAX 2000 IPS provides a dual CPU system with two MP cards. When Emergency Notification from hardware is detected, the changeover from an active MP card to a standby MP card will occur. If the active MP card becomes out of order for any reason, the standby MP card starts up automatically. The standby MP card employs a Cold Standby System that will restart initialization by the changeover from the active MP card.

Bandwidth Control
This feature allows to assign an available bandwidth threshold for VoIP traffic within a Location and between Locations, and to restrict outgoing/incoming calls when the VoIP traffic exceeds the threshold. The Location is a group of VoIP devices (IP Enabled Dterm, IP-PAD, or Peer to Peer IP trunks (built-in IP trunks)), which the same VoIP communications parameters such as codec selection list and ToS field value are assigned.

When the VoIP traffic over CCIS exceeds the threshold, the call can be routed to legacy trunks (TDM network). When exceeding the threshold, the system can store fault information and provide external alarm indication.

Boss / Secretary Calling
A secretary with a Multiline Terminal can use an appearance of the boss' extension to screen calls for that extension, and announce and/or transfer calls to that extension. Additionally, the secretary can call the boss during a busy condition and send a Message Waiting Indication to the boss' station.

Broker's Call This feature allows a Multiline Terminal or Single Line Telephone user to alternate between two parties, talking to one party while the other party remains on Hold on the same line. The Multiline Terminal user uses the TRF or ANS key to alternate between the two parties. The Single Line Telephone user uses the Hold feature to alternate between the two parties. **Call Back** This feature allows a calling party to set an automatic Call Back when a busy or no answer condition is encountered. When the busy station becomes idle, the station that set the Call Back will be called. In case of Call Back no answer, the Call Back to the setting station is initiated immediately after the called station goes on hook after making a call or accessing a feature.	**Call Forwarding** Call Forwarding allows calls directed to a station to be routed to another station, an Attendant, an outside number or voice mail equipment. The types of Call Forwarding provided are: Attendant Call Forwarding Setup and Cancel Call Forwarding - All Calls Call Forwarding - Busy Line Call Forwarding - No Answer Call Forwarding - Destination Multiple Call Forwarding - All Calls Multiple Call Forwarding - Busy Line Multiple Call Forwarding - No Answer Split Call Forwarding - All Calls Split Call Forwarding - Busy Line Split Call Forwarding - No Answer Call Forwarding – Logout (Dterm IP) Call Forwarding - Override Group Diversion **Attendant Call Forwarding Set-up and Cancel** All of the various types of Call Forwarding can be set up or canceled from both Attendant Consoles. **Call Forwarding - All Calls** This feature allows all calls directed to a particular extension to be rerouted to an alternate destination, regardless of the busy or idle status of the extension. Call Forwarding - All Calls can be set by an Attendant Console, the individual station user, a Multiline Terminal with a secondary appearance of the station's extension, or from another station (which can program itself to be the destination of the rerouting).

Call Forwarding - Busy Line

This feature permits a call to a busy extension to be routed to a pre-designated station, Attendant Console, or voice mail equipment. Call Forwarding - Busy Line can be set or canceled by an Attendant Console, the individual station user, or a Multiline Terminal with a secondary appearance of the station's extension.

Call Forwarding - No Answer

This feature reroutes calls to extensions which do not answer. These calls can be rerouted to another station, an Attendant Console or voice mail equipment. Call Forwarding - No Answer can be set by the individual station user, an Attendant Console, or by a Multiline Terminal with a secondary appearance of the station's extension.

Call Forwarding - Destination

This feature allows a station (A) user to set Call Forwarding - All Calls from another station (B) within the system, to the user's station (A).

Multiple Call Forwarding - All Calls

When a forwarded call is rerouted to a station that has also set a Call Forward, the call can be forwarded to another station. A call can be forwarded up to a maximum of five times, as specified in system programming.

Multiple Call Forwarding - Busy Line

This feature permits a call to a busy station to be forwarded, multiple times, to a pre-designated idle station.

Multiple Call Forwarding - No Answer

This feature permits a call to an unanswered station, the ability to be forwarded multiple times to a pre-designated station that does not have Call Forwarding - No Answer set or to the Attendant Console.

Split Call Forwarding - All Calls

This feature allows all internal and external calls to a busy extension to be rerouted to different destinations individually, regardless of the busy or idle status of the extension.

Split Call Forwarding - Busy Line

This feature allows internal and external calls to a busy extension to be rerouted to separate destinations. Destinations may be an internal station, Attendant Console, or voice mail.

Split Call Forwarding - No Answer

This feature allows internal and external calls, to extensions that do not answer, to be rerouted to separate destinations individually.

Call Forwarding – Logout (Dterm IP)

This feature allows a call terminated to an IP station in logout status to be forwarded to a predestinated station, outside number, Attendant Console or Digital Announcement Trunk (DAT). This feature is also applicable to the IP stations when the LAN cable is pulled out or the power is off.

Call Forwarding - Override

This feature allows the call forward destination station to call the station which set call forward. The call forward setting will be ignored.

Group Diversion

This feature allows all calls terminated to an extension that are not answered within a predetermined time to be forwarded to a pre-designated station.

Call Park

This feature enables a station user or attendant to place a call into pre-designated Call Park locations. The station user or attendant is then free to process other calls. This feature is available system wide and for individual tenants.

Call Park - System

When a call is parked by Call Park-System, the call can be retrieved from Call Park by any station in the system.

Call Park - Tenant

When a call is parked by Call Park - Tenant, the call can be retrieved from Call Park-Tenant by any station within the tenant from which the call was originally parked.

Call Pickup

This feature enables a station user to answer any call directed to another station, to a station within the user's own Call Pickup Group, or to a station within a different Call Pickup Group. Three Call Pickup methods are available: Call Pickup - Direct, Call Pickup - Group, and Call Pickup - Designated Group.

Call Pickup - Direct

This method permits a station user to pickup a call to any other station in the system by dialing a specific Call Pickup feature access code and the number of the called extension.

Call Pickup - Group

This method permits a station user to answer any calls directed to other extensions in their preset pickup group by dialing a Call Pickup - Group feature access code.

Call Pickup - Designated Group

This method permits a station user to answer an incoming call directed to another group by dialing the Call Pick-up - Designated Group feature access code and any station within the group to which the ringing station belongs.

Call Redirect

Without answering incoming calls or held calls that terminate to the line keys of a Multiline Terminal, the calls can be transferred to a pre-programmed station or Voice Mail System. Two transferring destination number can be designated per tenant, in system data programming. This feature can be used together with the Caller ID Display feature.

Call Transfer

This feature permits a station user to transfer a call to another station in the system directly, or with assistance from the attendant.

Call Transfer – All Calls

This feature permits a station user to transfer incoming or outgoing calls to another station within the system without attendant assistance.

Call Transfer – Attendant

This feature permits a station user, while connected to an internal or outside call, to signal the Attendant and have the Attendant transfer the call to another station within the system or to an outside connection.

Caller ID Class

This feature receives the calling subscriber's name and number sent from a public network using a MODEM signal and displays the name or number on an LCD of a Multiline Terminal and Attendant Console.

Caller ID Display

Without answering incoming calls or held calls which terminate to the line keys of a Multiline Terminal, the calling party's information can be confirmed by the indications on the LCD.

Caller ID Station

This feature enables a user to connect an analog telephone with Caller ID display function, and provides the calling party's number and name on the display without answering incoming calls.

Camp-on
This feature provides selected stations or outside calls with Camp-On capability to a busy internal station. Two Camp-On methods are provided. The call waiting method allows a station or an outside party to camp itself on to a busy station. The transfer method allows a transferred outside call to be camped-on to a busy station.

Centrex Compatibility
A combination of features allows full integration of the *UNIVERGE* NEAX 2000 IPS with Centrex service.

Check In / Check Out
When this feature is activated, the following operations occur:

- Check In
  Room Cutoff is cleared.
- Check Out
  Room Status printout is supplied.
  Do Not Disturb is reset.
  Room Cutoff is set.
  Message Waiting is reset.
  Automatic Wake Up is cleared.

CID Call Back
When an incoming call is terminated from station and trunk with Caller ID (calling number information), and called station does not answer, the calling number is registered to the system memory and MW lamp is lit. After the calling number is registered, by operating from station, confirmation / delete / call back is available. Besides, when the called station answers, the calling number is also registered to the Last Number Redial memory. The station can search and call back to that number by the operations of Last Number Redial / Stack Dial feature.

CID Call Routing
This feature allows designating a call terminating system based on the calling party number received from the network.

Class of Service
This feature permits all stations to be assigned a Class of Service in accordance with the degree of system use desired. The Class of Service is used to assign restrictions for trunk access and feature access.

Code Restriction
This feature allows the *UNIVERGE* NEAX 2000 IPS to be programmed to restrict outgoing calls according to specific area and/or Central Office codes. This restriction is controlled on the basis of a three-digit area code or six-digit area and office code numbering plan.

Conference (Three/Four Party)
This feature provides a station user the ability to add-on another party (trunk or station) to a call already in progress. Single Line Telephone users can add up to one additional party and Multiline Terminal users can add up to two additional parties.

Conference (Six/Ten Party)
This feature permits a station user or Attendant (conference leader) to establish a Conference among as many as six or ten parties (including the Conference leader).

Conference (32 Party)
This feature permits a station user, Attendant, or a trunk party to establish a conference among as many as 32 parties (including the conference leader). Two Conference methods are available: Group Call and Meet-Me Conference.

Group Call
This feature enables a station user (PS, Multiline Terminal, Single Line Telephone) within the system or a trunk party to establish a conference among as many as 32 parties. It also enables a station user to page a maximum of 31 parties simultaneously, excluding the conference leader. Three Group Call methods are available: Group Call - Automatic Conference, Group Call - Broadcasting, and Group Call - 2 Way Calling.

Group Call - Automatic Conference
This feature enables a station user to establish a conference among as many as 32 parties. From a station or Attendant, a maximum of 31 stations/trunks can be paged simultaneously except the conference leader.

Group Call - Broadcasting
This feature enables a station user to page a maximum of 31 parties simultaneously except the group call leader. After paged parties answer, the leader can speak to the paged parties (the paged parties only hear the leader's voice).

Group Call - 2 Way Calling
This feature enables a station user to page a maximum of 31 parties simultaneously except the group call leader. After one of the paged parties' answers, paging becomes 2-way calling between the leader and the first answered party and automatically stops paging other parties.

Meet Me Conference
This feature enables station users (PS, Multiline Terminal, Single Line Telephone) within the system, Attendants, or trunk parties to join a conference of as many as 32 parties by dialing a specific access code. The conference participants are automatically connected to the conference trunk. Conference participants may call in at preset time or may be directed to do so by a conference coordinator.

Consecutive Speed Dialing
For Speed Dialing, all digits are registered as a Speed Dialing Code. In the case of Consecutive Speed Dialing, the common portion of the number is registered as a speed calling code, and the remaining digits of each number are dialed by each individual calling station or by using a Station Speed Dial key on a Multiline Terminal.

Consultation Hold
This feature permits a station user to hold any incoming or outgoing CO call, tie line call, or any intra-office call while originating a call to another station user within the system.

Customer Administration Terminal (CAT)
In addition to the Maintenance Administration Terminal (MAT), programming can be done from selected Multiline Terminals with LCD. The designated Multiline Terminals can be placed in program mode, and system data can then be changed. To prevent unauthorized changes, password levels are assigned, providing authorization for access to certain areas of programming.

Data Line Security
This feature allows line circuits that are used for data transmission to be protected from interruptions such as Attendant Camp-On, Executive Override, and Attendant Override.

Delayed Ringing
This feature enables trunks and station lines to ring immediately at the terminating station, but also, after a programmable period of time has elapsed, to ring at secondary Multiline Terminals with that trunk or line appearance.

Diagnostics
To assist maintenance personnel, the *UNIVERGE* NEAX 2000 IPS provides diagnostic capabilities such as fault code generation, device status information and alarm information recording which can be accessed from the Maintenance Administration Terminal (MAT) or Customer Administration Terminal (CAT).

Dial By Name
This feature allows a Multiline Terminal user to search for a desired number by name. The number and name are registered in the system and they are shown on Multiline Terminal LCD. The Multiline Terminal user can search for the desired number by name using up or down soft keys. When the Multiline Terminal user finds the desired number, the call can be originated by pressing the Line/Trunk key or going off hook.

Dial Conversion
The system can be assigned to use rotary Dial Pulse (DP) or Dual Tone Multi-frequency (DTMF) trunks and stations. This feature provides for the repeating of digits dialed by the station user onto the C.O. trunks.

Direct Data Entry
This feature allows a maid or other hotel personnel to enter numeric data to the Property Management System (PMS), using the guest room station for entry through dial operation. The same numerical data can be output to a Hotel/Motel Printer by system data programming.

Direct Digital Interface
This service feature provides the capability to connect trunks from the *UNIVERGE* NEAX 2000 IPS directly to T1 carrier links using either a private or public network.

Direct Inward Dialing (DID)
This feature provides for incoming calls from the exchange network (except FX or WATS) to reach any station within the system without attendant assistance.

DID Call Waiting
This feature allows an incoming call on a DID trunk or a tie line to automatically be Camped-On to the destination station if the destination station is busy.

DID Digit Conversion
This feature allows the *UNIVERGE* NEAX 2000 IPS to convert the digits received from the serving C.O. to valid station numbers when the C.O. numbering plan differs from the desired station numbering plan.

DID Name Display
This feature allows name assignment for a DID number received from a public network, and displays the name on an LCD of a Multiline Terminal or Attendant Console.

Direct Inward System Access (DISA)
This feature allows an outside caller to access the system using an exchange network connection without Attendant or station assistance. The outside user may originate calls over any or all of the system's facilities such as WATS, FX, Tie Line or CCSA. The outside user can also directly call stations and access miscellaneous trunks for such features as dictation access.

Call Forwarding Set by DISA
This feature allows an outside caller to set Call Forwarding - All Calls by using Direct Inward System Access (DISA) code.

Direct Inward Termination (DIT)
This feature automatically routes incoming network exchange calls directly to a pre-selected station without Attendant assistance. The call can then be processed by the called party. Three-party Conference, Call Transfer, etc., are handled in the same manner as any normal trunk call.

Direct Outward Dialing (DOD)
This feature permits any station user the ability to gain access to the exchange network by dialing an access code and receiving new dial tone. The user may then proceed to dial the desired exchange network number.

Direct Station Selection/Busy Lamp Field (DSS/BLF) Console
This feature allows a DCU-60-1 unit associated with a legacy Multiline Terminal to be used as a Direct Station Selection/Busy Lamp Field (DSS/BLF) Console. When the buttons on the DCU-60-1 unit are programmed for Direct Station Selection (DSS) buttons, up to 60 stations can be directly accessed in addition to those already appearing on the Multiline Terminal. Busy status for each station is indicated by a red LED associated with each button. In addition, the DSS console can provide the following functions:

- Message Waiting - Set/Cancel/Status Display
- Do Not Disturb - Set/Cancel/Status Display
- Automatic Wake Up No Answer - Status Display/Cancel
- Agent Busy Out - UCD - Status Display
- Line Lockout - Status Display
- Room Cutoff - Set/Cancel/Status

Busy Out Status Console
This feature allows a DCU-60-1 unit associated with a Multiline Terminal to be used as a Busy Out Status Console. This feature is activated by use of a Function Mode key on a DSS/BLF Console. Busy Out Status for each station is indicated by a red LED associated with each button.

Do Not Disturb Console
This feature allows a DCU-60-1 unit associated with a Multiline Terminal to be used as a Do Not Disturb (DND) Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. DND set status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/cancel the DND status of other stations using the DND Console.

Message Waiting Console

This feature allows a DCU-60-1 unit associated with a Multiline Terminal to be used as a Message Waiting (MW) Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. The Message Waiting status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/reset MW status using the MW Console.

Room Cutoff Console

This feature allows a DCU-60-1 unit associated with a Multiline Terminal to be used as a Room Cutoff Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. The Room Cutoff status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/cancel Room Cutoff to another station using the Room Cutoff Console.

Wake Up No Answer Console

This feature allows a DCU-60-1 unit associated with a Multiline Terminal to be used as a Wake Up No Answer (WU) Console. This feature is activated by a function mode key on a DSS/BLF Console. The No Answer status for each station is indicated by a flashing green LED associated with each button.

Distinctive Ringing

This feature provides Distinctive Ringing patterns to the station so that the station user can distinguish between internal and external incoming calls. This feature also enables the LED associated with the line key of the Multiline Terminal to flash in two colors according to the kind of incoming call.

Do Not Disturb

This feature restricts incoming calls to a station and can be set by an individual station or from the Attendant Console. Placing a station in Do Not Disturb (DND) does not prevent a station from originating a voice or data call or from receiving a data call. This feature also allows a station to ensure privacy from telephone interruptions while on an outgoing call. Additionally, the Attendant Console can place a group of stations in the Do Not Disturb condition.

Do Not Disturb – Group

This feature allows the system to schedule to set/cancel Do Not Disturb for a group of stations at appointed time. The system has up to four patterns of timetable, and each timetable has time to set/cancel Do Not Disturb in a day (the time is programmable by the system). The timetable can be assigned for day of the week. The different timetable can also be assigned for specific dates of the year.

Do Not Disturb - Hotel/Motel

This feature allows the Attendant Console(s), Hotel/Motel Front Desk Instrument(s), guest stations or Property Management System (PMS) terminal(s) to place individual stations into Do Not Disturb. Calls can be placed from stations set in DND.

Do Not Disturb-System

This feature simultaneously restricts incoming calls to a pre-assigned group of stations by operation from the Hotel/ Motel Front Desk Instrument(s). Attendant Console(s) and Hotel/Motel Front Desk Instruments can use the DND OVR key to override this Do Not Disturb setting.

Dterm Assistant Dterm Assistant is Web-based software, which resides on the server and provides end users with the ability to maintain their Dterm Multiline Terminals and the 2000 IPS telephony features such as Speed Dialing from their Web-enabled PCs. The Dterm Assistant operates in a client – server environment and can manage multiple 2000 IPS systems over a Local Area Network (LAN)/Wide Area Network (WAN). **DtermIP** DtermIP is an IP-based Multiline Terminal, which provides a built-in capability of peer-to-peer IP communications. The *UNIVERGE* NEAX 2000 IPS system provides the DtermIP with same IP communications capabilities of an IP Enabled Dterm (The IP Enabled Dterm is a Dterm Multiline Terminal with an add-on IP adapter unit). **Elapsed Call Timer** This feature provides a display of the elapsed time while a Multiline Terminal with LCD is connected to any trunk. **Enhanced 911** This feature allows the PBX to transmit a caller's emergency service identification information to an Enhanced 911 Emergency system. **Executive Calling** This feature allows a station to be assigned a VIP class. This provides special ringing to a called station when that station is idle, and automatic sending of three tone bursts to a called station when that station is busy, provided the call was originated from a station assigned as VIP class.	**Executive Override** This feature allows selected users to override a busy condition on a called station. A warning tone is transmitted to both stations in the busy call before the busy condition is overridden, and a three-party Conference is then established. **External Paging with Meet-Me** This feature allows a station user or attendant dial-access to local voice paging equipment and connects both parties automatically after the paged party has answered the page by dialing an access code. **Fax Arrival Indicator** When a call from a C.O. line (Direct-Inward-Termination, Direct-Inward-Dialing, Automated Attendant), station or tie line has terminated to a facsimile machine, a related lamp on a pre-designated Multiline Terminal is caused to light, indicating reception of a facsimile call. **Fax over IP** This feature allows the system to transmit facsimile communications over IP network, via Local Area Networks (LAN) and corporate Wide Area Network (WAN). Since PBX regards facsimile equipment as one of ordinary telephones, IP Packet Assembler/Disassembler (IPPAD) and Voice Compression Trunk (VCT) are required for facsimile uses over IP network same as legacy stations. The facsimile transmission procedure (T.30 or G.711/G.726 pass-through) is supported with IP-PAD/ VCT.

Feature Activation from Secondary Extension This feature allows the Multiline Terminal user to access an appearance of another extension and program certain features from that extension. **Flexible Line Key Assignment** Multiline Terminals can have any desired line-key assignment. This feature permits assignments to be tailored to each individual's needs. (The terminal's primary extension line appearance is the only line key that cannot be reassigned.) **Flexible Numbering Plan** The *UNIVERGE* NEAX 2000 IPS has a Flexible Numbering Plan. All access codes and station numbers and can be assigned in system programming. Refer also to the Single Digit Dialing Features and Specifications, which further increases the flexibility of the system. **Flexible Ringing Assignment** This feature allows lines on Multiline Terminals to be individually programmed to ring or not ring. **Forced Account Code** This feature forces the user to enter an Account Code (up to 8 or 10 digits) for all outgoing calls. The Account Code must be dialed before dialing the outgoing number. Calls are processed only when the dialed Account Codes are valid.	**Group Call - Automatic Conference (6/10 Party)** This feature allows a Multiline Terminal user or single line telephone user within the system to establish a conference among as many as six or ten parties. From a Multiline Terminal /Single Line Telephone, a maximum of 9 stations can be paged simultaneously plus the originator. The stations are assigned to the simultaneous paging groups as participants by the system data beforehand. **Group Call - 2 Way Calling** This feature allows a Multiline Terminal/Single Line Telephone to page a maximum of fifteen parties simultaneously including the originator. After one of paged parties answers, the paging becomes the 2 Way Calling between the originator and the first answered party, automatically stops paging other parties. The stations are assigned to the simultaneous paging groups as participants by the system data beforehand. **Group Call by Pilot Number Dialing** This feature allows a station user (Multiline Terminal / Single Line Telephone / PS) or a trunk party to page a group of stations simultaneously by dialing a pilot number. The maximum of 32 stations can be assigned to a paging group, and the paging group is associated with the pilot number. After one of the paged stations answers, the paging becomes a 2 way calling between the calling party and the first answered station and automatically stops paging other stations. **Group Listening** When a Multiline Terminal user makes a call using the handset, pressing the SPKR key will allow others to listen through the built-in speaker of the Multiline Terminal. The user may continue talking on the handset at the same time.

Hands-free Answerback
This feature allows the station user to answer a voice call without lifting the handset.

Hands-free Dialing and Monitoring
This feature allows the station user to dial or monitor a call without lifting the handset.

Hold
This feature permits a user to Hold a call in progress. After Hold has been set, the station user can make or answer new calls.

Call Hold
This feature permits a user to Hold a call in progress by sending a hookflash and dialing the Call Hold feature access code, or by pressing the Call Hold key. This line can then be used for originating another call or returning to a previously held call.

Dual Hold
This feature permits a station user who is placed on Hold by another station to place that station on Hold also.

Exclusive Hold
This feature allows a Multiline Terminal user to place a call on Hold and to exclude all other station users from retrieving the held call.

Non-exclusive Hold
This feature allows a Multiline Terminal user to place a call on Hold that may be retrieved by any station that has an appearance of the held line.

Hotel/Motel Attendant Console
The Attendant Console can be programmed to function as a Hotel/Motel Attendant Console. In addition to the business features and functions of the Attendant, the Hotel/Motel Attendant Console can set Room Cutoff (individual and group), Automatic Wake Up, Message Waiting, and Do Not Disturb (individual and group).

Hotel/Motel Front Desk Instrument
A Multiline Terminal with LCD can be programmed to function as a Hotel/Motel (H/M) Front Desk Instrument. This can be used to set and cancel standard H/M features such as Message Waiting, Do Not Disturb, Automatic Wake Up, and Room Cutoff.

Hotline – Inside/Outside
This feature causes the terminal to place a call to another station or to an outside party automatically when the user selects the Hotline extension.

House Phone
This feature allows selected stations to reach the Attendant simply by going off-hook.

Individual Attendant Access
This feature permits a user to call a specific Attendant by dialing an Attendant call code.

Intercept Announcement
This feature provides the automatic interception of Direct Inward Dialing (DID) and Tie Line calls which cannot be completed due to unassigned station or level. The caller hears a recorded Intercept Announcement that informs the caller that an inoperative number was reached, and may supply the number for information.

Intercom
Three types of Intercoms are available: Manual Intercom, Automatic Intercom, and Dial Intercom. Each type of Intercom provides access to a small group of Multiline Terminals with simplified calling methods.

Manual Intercom

The Manual Intercom groups have up to six Multiline Terminals sharing a common signal path. Users can call other members of the Manual Intercom group by pressing a Manual Intercom key; each press sends a tone burst over the speakers of all the terminals in the group. When another user answers the call, a speech path is activated.

Automatic Intercom

Automatic Intercom provides a path for Voice Announcement Calls with Handsfree Answerback between two Multiline Terminals using a line key. Private conversations can be held by using the Multiline Terminal handsets. The Busy/Idle status of the associated Multiline Terminal is displayed on the Automatic Intercom line key LED.

Dial Intercom

Dial Intercom comprises up to 10 Multiline Terminals which can call each other using a dedicated Dial Intercom line key with abbreviated dialing. Dial Intercom calls can be Voice Announce with Handsfree Answerback or ringing calls.

Internal Tone/Voice Signaling

Multiline Terminals can signal incoming internal calls by Voice Announcement or by ringing according to the programmed mode (Voice first or Ring first) of the called terminal. The caller can dial the digit 1 to change from Voice Announcement to Ring Tone or vice versa. The Multiline Terminal assigned this feature can program the following two modes:

• Voice Mode: allows an incoming call to terminate with Voice Announcement.
• Tone Mode: allows an incoming call to terminate with ringing.

Internal Zone Paging with Meet-Me

This feature allows the Attendant Console and system users to page over the built-in speakers of the Multiline Terminals within the assigned zone or all zones.

IP Enabled Dterm

This feature provides a Dterm Series E/i Terminal combined with an IP Adapter unit with the capability to connect *UNIVERGE* NEAX 2000 IPS via IP networking (IP Enabled Dterm). The IP Enabled Dterm provides users with all features currently available in Dterm Series E terminals.

Last Number Redial

This feature allows users to redial the last station-to-station or outside number they dialed using a feature access key or a feature access code. This is useful when the called station is busy or does not answer.

Least Cost Routing - 3/6 Digit

This service feature allows the *UNIVERGE* NEAX 2000 IPS to be programmed to route outgoing calls over the most economical facility (WATS, FX, DDD). Based on the individual area code and office code dialed (6-digit analysis), the system examines the programmed tables and uses the trunk in the order specified.

Line Lockout

This feature automatically releases a station from the common equipment if the station remains off-hook for longer than a programmed interval before dialing. Howler tone may be programmed to be sent to the station in Line Lockout.

Line Pre-selection

This feature provides the station user with two ways to select an idle, held, recalling, or ringing line before going off-hook.

Maid Status

This feature allows the Hotel/Motel (H/M) Front Desk Instrument, Property Management System (PMS) terminal, or guest room station (using special access code) to register the condition of each guest room.

Maintenance Administration Terminal
The Maintenance Administration Terminal (MAT) is a personal computer (PC) that provides an interface to the PBX via the *UNIVERGE* NEAX 2000 IPS CPU card. The MAT PC must have the MATWorX IPS program properly installed to communicate with the PBX. MATWorX is required for system software registration, activation and system data backups.

Message Center Interface (MCI)
This feature provides an interface with a customer supplied Voice Mail System (VMS) which can send Message Waiting lamp control data to the *UNIVERGE* NEAX 2000 IPS using an RS-232C interface.

Message Registration
This feature provides output from the system to a call accounting system using an RS-232C interface. This allows the Hotel/Motel clerk to retrieve the information needed to charge for local and toll calls.

Message Reminder
This feature allows a user or Attendant to turn on the message waiting (MW) lamp of a Single Line Telephone, or the Message Reminder (MSG) LED of a Multiline Terminal (if assigned).

Message Waiting
This feature allows the Attendant Console, Hotel/Motel (H/M) Front Desk Instrument, administrative station, or Property Management System (PMS) terminal to light a lamp (on an uninterrupted or interrupted basis) on a Single Line Telephone or Multiline Terminal to indicate a message is waiting. In addition to the lamp indication control, this feature also provides the Voice Message Waiting service that an originating station user can set to Message Waiting with a prerecorded message by using the Digital Announcement Trunk card (PN-2DATA).

Miscellaneous Trunk Access
This feature allows the connection of various types of external facilities. In addition to Loop and Ground Start Trunks, the following can also be interfaced with the *UNIVERGE* NEAX 2000 IPS: CCSA Lines Code Calling Equipment, Dictation Equipment, Foreign Exchange (FX) Lines, Radio Paging Equipment, and Wide Area Telephone Service (WATS) lines. Refer to separate features, Direct Inward Dialing (DID), and Tie Line Access for more applications of Miscellaneous Trunk Access.

CCSA Access
This feature allows connection to or from a Common Control Switching Arrangement (CCSA) network. A CCSA network is a special, privately-leased network constructed for one customer's exclusive use that replaces or augments the public switched network services.

Code Calling Equipment Access
Code Calling Equipment consists of external paging units and external dialers requiring dial access from the *UNIVERGE* NEAX 2000 IPS.

Dictation Equipment Access
This feature permits dial access to customer provided Dictation Equipment, and in some instances allows them to maintain telephone dial control of normal dictation system features.

Foreign Exchange (FX) Access
An FX line is a line that is extended and terminated at a distant Central Office. With this feature, outgoing calls over the FX line become a local call at the distant C.O.

Radio Paging Equipment Access
This feature provides station users dial access to Radio Paging Equipment, and to selectively tone - or voice/ tone-alert individuals carrying pocket paging devices. The paged party (when on premises) can be connected to the paging party by going to the nearest station and dialing an answer back code.

Wide Area Telephone Service (WATS) Access
This feature allows any station user direct dial access to outgoing WATS lines.

Mobility Access
This feature allows a user to make/receive telephone calls from a remote location using a Mobile/Cell Phone via the PBX, as if the user was at his/her desk phone.

Modem over IP
This feature allows the system to transmit modem communications over IP network, via Local Area Networks (LAN) and corporate Wide Area Network (WAN). IP Packet Assembler/Disassembler (IPPAD) and Voice Compression Trunk (VCT) are required for modem over IP network. The modem transmission procedure (G.711/G.726 pass-through) is supported using the 8IPLA/24IPLA IP-PAD only.

MP Program Download (FTP)
UNIVERGE NEAX 2000 IPS provides Online MP Program Download via IP network. The IPS downloads the MP upgrade program from FTP server using MATWorX. Immediate or scheduled changeover to the upgrade program is available. It is also possible to change back to the previous program that was in use before the changeover (changeback).

Multiple Language Display
A language displayed in the LCD of a Multiline Terminal and Attendant Console can be selected on a per-station basis or on a per-operator console basis. The available languages are the same as used on a system-wide basis.

Multiline Terminal Attendant Position
A Multiline Terminal with LCD can be programmed to function similar to an Attendant position. This Attendant position has limited access to Attendant related features and functions and can be substituted where an Attendant is required but an Attendant Console is not necessary. When an DCU-60-1 unit is associated with this Attendant Multiline Terminal enhanced operation is available.

Music on Hold
This feature plays music when a line is placed on hold. Music is provided by a circuit board memory chip, IP adpter, or a local music source, such as a CD player or a radio.

Night Service
This feature provides a variety of methods for handling incoming calls when the system is in night mode. These include:

- Attendant Night Transfer
- Call Rerouting
- Day/Night Mode Change by Attendant Console
- Day/Night Mode Change by Station Dialing
- Day/Night Mode Change by System Clock
- Night Connection-Fixed
- Night Connection-Flexible
- Trunk Answer Any Station (TAS)
- Overflow for TAS Queue
- Overflow Limit for TAS

Attendant Night Transfer

When the Attendant Console is in Night Service, any operator directed calls (dial 0 calls) are automatically routed to a preprogrammed station. Priority Calls and Off-Hook Alarms which terminate to an Attendant are also routed by this feature.

Call Rerouting

This feature provides flexible reroute capabilities for a variety of calls when the system is in night mode.

Day / Night Mode Change by Attendant Console

This feature provides activation of DAY/NIGHT Mode Change by depressing a predetermined key from the Attendant Console.

Day / Night Mode Change by Station Dialing

This feature allows selected stations to activate a change from day mode to night mode by dialing a special code.

Day / Night Mode Change by System Clock
This feature provides automatic activation of DAY/NIGHT Mode Change by using System Clock.
Night Connection - Fixed

This feature allows incoming calls normally terminated to the Attendant to reroute to a predetermined station when the system is placed in Night Service.

Night Connection - Flexible

This feature provides incoming calls normally terminated to the fixed night station to be Call Forwarded to another station.

Trunk Answer Any Station (TAS)

This feature allows any station, other than one with incoming restrictions, to answer incoming calls when the system is in the night mode. When this feature is activated, incoming exchange network calls will activate a common alert signal at the customer premises. By dialing a specified code, any station may answer the call and then extend it to any other station by means of the Call Transfer feature.

Overflow for TAS Queue

If the TAS call is not answered by predetermined time, the call will be forwarded to predetermined station/Attendant Console/Announcement Service.

Queue Limit for TAS

When a DID call is converted to TAS and the number of using lines reaches queue limit, this feature provides the system to restrict the next call terminating.

No CID Call Routing

This feature allows designating a call terminating system based on the reason for the absence of calling party number received from the network.

Off-hook Alarm

This feature allows a station user to call the Attendant, or a pre-designated station, by simply staying off-hook for a preprogrammed period of time. The calling number is automatically displayed at the Attendant Console, or the pre-designated station if equipped with an LCD.

Off-Premises Extensions

This feature allows the connection of a single line telephone in an off-premises location. The connection to the Off-Premises Extension can be through direct copper or through the local telephone company.

Open Application Interface (OAI)

Provides a computer-to-PBX interface, allowing a computer to control the function of the *UNIVERGE* NEAX 2000 IPS. The IPS can be customized to accommodate most customer applications. Application software can be provided by NECAM, an outside software house, or a customer.

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Optical Interface
The Optical Interface Card (PN-M10) provides an internal optical fiber modem to the T1/E1 or Remote PIM network. The PN-M10 provides a bandwidth of 6.3 Mbps and supports a maximum distance of 10 kilometers (or 6.25 miles) point to point. The *UNIVERGE* NEAX 2000 IPS only supports Single Mode optical fiber

Pad Lock
This feature temporarily restricts telephones from making unauthorized calls by dialing special access code when station users are away from their seats.

Periodic Time Indication Tone
This feature provides a periodic tone to the station user who has made an outgoing call. This feature can be allowed or denied for each station.

Pooled Line Access
A line key can be assigned to access Pooled Lines. Each line key will allow incoming, outgoing, or both-way access to a trunk route.

Power Failure Transfer
This feature provides for specified trunks to be automatically connected to designated Single Line Telephones in the event of AC power loss. It is normally used when the system is not equipped with reserve power.

Priority Call
This feature allows the Attendant to answer a call before other calls, at the Attendant's discretion.

Privacy
This feature restricts Multiline Terminal users from depressing a busy line button and entering a conversation unless permitted by the Multiline Terminal user currently on that line button or if the line button is assigned for Direct Privacy Release.

Direct Privacy Release
This feature allows a station user with a secondary appearance of another extension in the system to access that extension when it is being used by someone else. This feature allows for a simplified method for establishing a conference. In addition, this feature can be used to emulate PC dialing, where a single line extension connected to a PC can appear on a Multiline Terminal and be accessed by the Multiline Terminal user after the PC is completed dialing.

Manual Privacy Release
This feature allows a Multiline Terminal user to enter a conversation on a busy line button if the Multiline Terminal user already in the conversation allows them by releasing Privacy.

Private Lines
Only a C.O. trunk assigned to that specific station is seized when a station user originates an outgoing C.O. call or when an incoming C.O. call is terminated at the station designated by Direct-In-Termination. In this manner, stations and C.O. trunks are to be associated on a 1-to-1 basis.

Property Management System Interface
The *UNIVERGE* NEAX 2000 IPS provides a data interface to a locally provided Property Management System (PMS). This enables communication between the *UNIVERGE* NEAX 2000 IPS and the PMS in order to provide computer control of Hotel/ Motel features.

Proprietary Multiline Terminal
The are several Multiline Terminals available which can be used with the *UNIVERGE* NEAX 2000 IPS.

- Legacy: Dterm Series i
- Legacy: Dterm Series E (Display/Non-Display)
- Dterm IP: Dterm Series E/Series i (Display)
- Legacy: Electra Elite(Display/Non-Display)

Automatic Idle Return

This feature returns a station to the idle state after 3 seconds of reorder tone is received due to the distant end disconnecting.

Called Station Status Display

This feature provides a display on the status of a called station on the LCD of the calling Multiline Terminal.

Calling Name and Number Display

This feature provides a display on the LCD of the Multiline Terminal receiving a call, indicating the station number/trunk number of the incoming call.

Group Feature Key

A station user can monitor the status of group member stations such as idle, busy, hold and ringing by lamp indication of the multiple line keys. If a line is idle, a station user can make a call to the member station by pressing the associated line key (instead of pressing DSS key). If a line is ringing or is on hold, a station user can pick up the incoming call or the call on hold by using My Line, and the ringing or held line is made idle.

Dynamic Dial Pad

This feature allows a user to make an outgoing call by first pressing a key on the keypad of Multiline Terminal without pressing the Speaker key or going off-hook.

Handsfree Unit

The built-in Handsfree Unit enables full Handsfree operation for both internal and external calls (No optional Handsfree Unit is required).

I-Hold / I-Use Indication

Multiline Terminals provide indication of which line keys have been placed on Hold, or are in use by that Multiline Terminal. The LED associated with the line key will give the appropriate indication.

Microphone Control

All Multiline Terminals are equipped with a Microphone Control button with an associated LED.

Multiple Line Operation

This feature allows for the appearance of multiple lines on the Flexible Line Keys and feature keys of all Multiline Terminals.

Mute Key

This feature allows the distant extension user, of a station user that presses a mute key during conversation, not to hear the station user's voice though the station user can hear the distant extension user's voice. By pressing the mute key again, the mute status returns to original conversation.

My Line Number Display

This feature allows the display of the Prime Extension in the LCD display of the Multiline Terminal during an Idle state.

Preset Dialing

This feature allows a Multiline Terminal user to prepare and verify a number in the display on the LCD before dialing. When a wrong number is entered, the user can correct the number before originating the call.

Prime Line Pickup

This feature allows a Multiline Terminal user to go off hook and originate a call from the line assigned as the Prime Line without depressing the associated line key.

Recall Key

Each Multiline Terminal is equipped with a Recall Key that is used to generate a hookflash to access features provided by the outside exchange, or to abandon a call while retaining the line for origination of another call.

Relay Control Function Key

This feature provides a Multiline Terminal with the ability to activate/deactivate relays (on a PN-DK00) to control external devices.

Ring Frequency Control

The ring frequency of the Multiline Terminal can be controlled on a station basis in system programming (eight ro fourteen frequencies are available) or by use of a function key on the Multiline Terminal.

Ringing Line Pickup

This feature provides the ability to answer any call ringing into a Multiline Terminal by just lifting the handset.

Soft Keys

According to the status of the Multiline Terminal, function keys (Soft Keys) are displayed in the third line on the LCD. If the status of Multiline Terminal changes, the Soft Keys will change automatically. Also if the Help key is pressed, explanation of indicated Soft Keys are shown on the LCD.

Volume Control

Multiline Terminals are equipped with common Volume Control keys for:

- Built-in Speaker / Handset Receiver Volume
- Ring Volume
- C.O. Transmission Level
- LCD Contrast
- Ring Tone Frequency

Remote Hold

This feature allows a Multiline Terminal user or an attendant to place a call on hold on the line button of the destination terminal, by pressing the Hold key.

Remote PIM over IP

When IPS DMR or 2000 IPS PIM are installed at remote site, and connected to a 2000 IPS or IPS DM at main site over IP network, the Main Site system controls and maintains the remote DMR or IPS PIM operation as one single system. If a communication failure occurs between the Main Site and Remote Site, the Remote Site automatically changes over to a survival mode and operates as a stand-alone system.

IPS DMR: IPS Distributed Model Remote (with CP31-A/B/C)
IPS DM: IPS Distributed Model (with CP24-A/B/C)

Reserve Power

This feature provides backup power from a 24V battery source in the event of a commercial power failure.

Resident System Program

This feature provides the installers a simple procedure to have the system generate system data according to the system hardware configuration, thereby providing immediate operation and shorter programming time. When activated, the system scans hardware configuration (such as line/trunk card slot location) and assigns system data (such as extension numbers, trunk numbers, etc.) according to a predetermined generic program assignment.

Return Message Schedule Display

This feature permits any station user to register his Return Schedule from his phone when he leaves his desk or the premises, and have the Return Schedule displayed on a calling Multiline Terminal with a Liquid Crystal Display (LCD) during his absence.

Room Cutoff

This feature allows the Attendant Console, Hotel/Motel (H/M) Front Desk Instrument, or Property Management System (PMS) terminal, or guest room telephones using a special access code, to temporarily restrict guest room telephones from making unauthorized calls when guests are away from their room, and when rooms are in Check Out status.

Room Status

This feature provides the Hotel/Motel (H/M) Front Desk Instrument with a visual display of the guest's room status. A supplementary print out (individual and summary) can be provided.

Route Advance

This feature automatically routes outgoing calls over alternate facilities when the first choice trunk group is busy. Users select the first choice route by dialing the corresponding access code, and the equipment then advances through alternate trunk groups only if the first choice is busy.

Save and Repeat This feature allows a Multiline Terminal to save a specific dialed number and then redial that number at a later time. **Security Alarm** This feature provides an indication on the Attendant Console when a contact closure occurs. **Set Relocation** This feature enables two stations to be moved from one location to another without reprogramming station data at MAT. **Single Digit Dialing** This feature provides the station user the ability to dial single digit codes to access certain features while still allowing the same digit dialed to be used as the first digit of station number. **Single Digit Feature Access Codes** This feature allows stations to access certain other system features by the direct dialing of a Single Digit Access Code, while receiving Busy Tone or Ring Back Tone. **SIP Trunk Interface** 2000 IPS supports Session Initiation Protocol (SIP) based connections **SNMP** Simple Network Management Protocol (SNMP) is a standard protocol for TCP/IP network management, which enables a network management application software to query a management agent (network device such as router, PC host, and hub) using a supported MIB (Management Information Base). The MIB is a database of network performance information that is stored on the network devices. The *UNIVERGE* NEAX 2000 IPS can support the SNMP standard MIB (MIB-II, defined in IETF RFC 1213) and private MIB and TRAP. This feature also enables the network management system (SNMP manager) to manage the IPS via Network Address Translation (NAT).	**Software Line Appearance (Virtual Extensions)** This feature permits assignment of circuits which do not physically exist, to be used as secondary extensions on Multiline Terminals. There are a possible 1020 virtuals and can be calculated by 1020 – number of Multiline Terminals (Dterm/DtermIP)=Total Virtuals available. **Stack Dial** This feature enables a Multiline Terminal or an Attendant Console to remember the numbers dialed in the preceding five calls, including the last number dialed. The stack dial numbers are sequentially displayed on the LCD display, thus allowing the station user to make an outgoing call by selecting the desired dialed number from the display. **Station Hunting** Three Station Hunting arrangements are available. Station Hunting - Circular processes the call no matter which station in the hunt group is called. Station Hunting - Terminal initiates a hunt only when the pilot number of a hunt group is called. Station Hunting - Secretarial is initiated when a busy station in a Station Hunting - Circular group or Station Hunting - Terminal group is reached. **Station Hunting - Circular** When a busy station in a hunt group is called, this feature permits the call to be processed automatically through the hunt group in a preprogrammed order from that station's position within the hunt group. **Station Hunting - Terminal** When a pilot number is dialed and that number is busy, sequential Station Hunting will be initiated. However, if a number other than the pilot number is dialed and that number is busy, busy tone will be provided rather than initiate Station Hunting.

Station Hunting - Secretarial

This feature allows assignments to be given to members of Terminal and Circular Hunting groups to reroute calls (when their hunting group is all busy) to a back-up hunting group.

Station Message Detail Recording (SMDR)

This feature provides a call record for outgoing station-to-trunk calls, incoming trunk-to-station calls (including Data Call) and tandem calls. This facilitates cost control by identifying trunk use and misuse by individual stations. Station Message Detail Recording (SMDR) enables call billing to customers and clients, and provides a means for checking local telephone bills.

Station Speed Dialing

This feature allows a station user to dial frequently called numbers by dialing an access code and an abbreviated code, or by depressing a feature key or line key assigned for Station Speed Dialing capability.

Step Call

This feature allows the Attendant or station user, after calling a busy station, to call an idle station by simply dialing an additional digit. This feature will operate only if the number of the idle station is identical to that of the busy station in all respects, except the last digit.

Supervisory Control of Peripheral Equipment

When various types of peripheral equipment (such as facsimiles, dictation equipment, Voice Mail, etc.) are connected to the line circuits of the *UNIVERGE* NEAX 2000 IPS, this feature allows the loop of the line circuit concerned to open for a programmable interval, and send a release signal to the peripheral equipment when the calling party disconnects.

System Clock Setup by Station Dialing

This feature enables a station user to set up the system clock, from Single Line Telephone, Multiline Terminal, and PS.

System Speed Dialing

This feature provides all users the ability to dial frequently called numbers using an abbreviated call code.

System Traffic

The *UNIVERGE* NEAX 2000 IPS is the stored program controlled digital electronic PABX employing PCM time division switching technique. Single stage time division switching is employed for all configurations of the IPS up to its maximum line and trunk configuration. Traffic capacity is expressed in Busy Hour Call Attempts (BHCA). This value is the number of call attempts that can be generated in the busiest hour of the day.

Tenant Service

This feature provides for more than one organization (tenant) to share the same *UNIVERGE* NEAX 2000 IPS system. Through system programming, each organization may be restricted to its own Central Office trunks, Attendant Consoles and extension group. In addition, incoming calls are directed to the specific tenant.

Terminal Login via NAT

The NEAX 2000 IPS supports network configuration via a router with NAT. NAT is a technology that translates an internal local IP address to a globally unique IP address before sending packets to the outside network. NAT is configured on the router at the border of the inside network and the outside network.NAT is used for efficient use of global IP addresses and for security.

Tie Lines

This feature allows any station user dial access or direct access to an E&M Tie Line.

Tie Line Tandem Switching

This feature allows trunk-to-trunk connections through the IPS without the need for any Attendant assistance or control. The major use of this feature is in association with a dial tandem tie line network to allow tie line connections and incoming tie line calls automatic access to, and completion of, local

	Central Office calls.
Timed Queue When a user originates an outgoing trunk call and the called party is busy or does not answer, the caller can set the Timed Queue feature. When this feature is set, the trunk seizure is repeated and the number is redialed after a predetermined time interval. **Timed Reminder** This feature allows the system to be programmed to automatically call stations at specified times. Upon answering, the station is connected to a recorded announcement or music source. **Trunk - Direct Appearances** This feature allows Multiline Terminal users the ability to access a CO line or E&M Tie Line without dialing an access code. For this feature, trunks must be assigned to the line keys on the Multiline Terminal. Incoming calls on CO lines can be answered on the appropriate trunk line appearance. **Trunk Queuing - Outgoing** This allows a station user, upon encountering a busy signal on a trunk, to dial a feature access code and enter a first-in, first-out queue. As soon as an outgoing trunk becomes available, stations in the queue will be called back on a first-in, first-out basis. **Trunk-to-Trunk Connection** This feature provides any station user with the ability to conference together two outside trunk calls and abandon the connection without dropping the Trunk-to-Trunk Connection. **Timed Forced Release** This feature will restrict Long duration calls by forcibly releasing the call after a preset time. The feature is available for outgoing, incoming and tandem calls.	**Uniform Call Distribution (UCD)** The Uniform Call Distribution (UCD) feature permits incoming calls to terminate to a prearranged group of stations. Calls are distributed in the order of arrival to idle terminals within the group, based on which terminal has been idle the longest period of time. Stations may log on/log off from the UCD group. Supervisor stations may monitor conversations of agents. **Busy In/Busy Out-UCD** This feature allows an agent in a UCD group to log their station onto or off of the group. This allows the system to control whether a call directed to the pilot number of the UCD group goes to that station or not. This prevents incoming calls from being directed to stations at which no agent is available. **Call Waiting Indication-UCD** This feature provides a visual indication when an incoming call to a UCD group is placed in queue, due to an “all agents busy” condition. An external relay controlled indicator or an LED on a Multiline Terminal can be used to provide Call Waiting Indication. **Delay Announcement-UCD** This feature allows the system to provide a recorded announcement to an incoming caller placed in queue to a UCD group. A single announcement, or two separate announcements, can be provided. **Hunt Past No Answer-UCD** This feature allows calls targeted at a UCD group to hunt past an agents station, after a no answer condition, if the agent forgets to log off of the group and the agent is unable (or not available) to answer the call.

Immediate Overflow-UCD

This feature allows a call directed to a UCD group to immediately overflow to another UCD group, upon encountering an “all agents busy” condition.

Priority Queuing-UCD

This feature allows the system to prioritize incoming calls by trunk route and on a per station basis, when the call enters a UCD queue. When a call is a considered as priority it is placed at the beginning of the queue.

Queue Size Control-UCD

On incoming DID/Tie line calls, the system can be assigned a threshold that limits the number of calls in queue. When the queue size threshold is exceeded, incoming callers are connected to busy tone.

Silent Monitor-UCD

This feature provides the UCD group supervisor with the ability to monitor a call to a UCD agent. The silent monitor function gives no indication (as an option) to either the agent or the calling party.

Uniform Numbering Plan (UNP) -Voice and Data

In the numbering plan for a network to be configured through the use of Tie Lines, a Uniform Numbering Plan (UNP) is employed. When UNP is employed, a station user from any PBX within the network can call a desired party by using a uniform dialing method based on the UNP.

Variable Timing Parameters

This feature gives the IPS the versatility to change timing duration using the Maintenance Administration Terminal (MAT) or the Customer Administration Terminal (CAT). All timing parameters are set initially in the Resident System Program. These timing parameters can be changed according to the customer's requirements.

Voice Guide

This feature provides a station user with an announcement that informs:

1. The result of the operation when the station user set or canceled the service feature, instead of service set tone.

2. Which service has been set to the station; such as, Call Forwarding - All Calls, Do Not Disturb or Message Waiting, when the station goes off-hook, instead of special dial tone.

Voice Mail Integration

This feature is used to interface the *UNIVERGE* NEAX 2000 IPS with a locally provided stand-alone type Voice Mail System (VMS). The VMS, connected to the using single line circuit (LC), is controlled by sending/receiving DTMF signals using this LC.

The VMS’s voice mail feature can be used by accessing this VMS directly from an extension. If a station sets its call forwarding destination to the VMS, calls to this station are connected to the VMS, and the messages can be registered according to the VMS instruction. In addition, the Message Waiting lamp of the station can be turned on automatically by the VMS.

Voice Mail Private Password

Voice Mail Password can be prevented from displaying in LCD of Multiline Terminals when connected to the Voice Mail System.

Voice Mail Transfer This feature has two functions that provide streamlined transfer access to voice mail. 1. One touch access to VMS: When an Attendant transfers an external call to a station, and if the station is busy or unanswered, the Attendant can transfer the call to a VMS by dialing "9" or by pressing a function key provided for this feature. 2. Transferring Camp-On call to VMS: When an Attendant sets Camp-On to a busy destination station for an external call, and if the destination station does not answer by predetermined time, the call can be automatically transferred to a VMS.	**Voice Over IP (H.323)** Provides the ability for the *UNIVERGE* NEAX 2000 IPS to connect via IP Trunk to other non-NEAX systems that are compliant with H.323. Allows Transparent Dialing and Calling Station Number Display between the systems. Gate keeper (GK1000) software and customer supplied server is required. **VoIP Log Collection** This feature provides Voice over IP (VoIP) log collection functions for analyzing VoIP related problems caused by IP device, usage method, network environment and other effects. A maintenance person will be able to understand its cause and figure out how to solve the problem. The following functions are supported. **Whisper Page** This feature allows a secretary to interrupt the boss in a private way. By pressing a feature key or dialing an Access Code, the secretary station can voice override the conversation between the boss and another party (station or trunk). When the conversation is interrupted, the boss can hear the secretary but the other party is unaware of the Voice Override.

CCIS Feature List

Attendant Camp-On with Tone Indication - CCIS	LDN Night Connection - CCIS
Attendant Controlled Conference - CCIS	Link Alarm Display - CCIS
Automatic Recall - CCIS	Link Reconnect - CCIS
Brokerage - Hot Line - CCIS	Message Waiting Lamp Setting - Attendant - CCIS
Busy Lamp Field (BLF) - CCIS	Message Waiting Lamp Setting - Station - CCIS
Busy Verification - CCIS	Miscellaneous Trunk Access - CCIS
Call Back - CCIS	Miscellaneous Trunk Restriction - CCIS
Call Forwarding - All Calls - CCIS	Multiple Call Forwarding - All Calls - CCIS
Call Forwarding - Busy Line - CCIS	Multiple Call Forwarding - Busy Line - CCIS
Call Forwarding - Don't Answer - CCIS	Multiple Call Forwarding - Don't Answer - CCIS
Call Forwarding - Intercept - CCIS	Multiple Console Operation - CCIS
Call Forwarding - Override - CCIS	Network Station Number - CCIS (FCCS)
Call Processing Indication - CCIS	Night Connection - Fixed - CCIS
Call Transfer - All Calls - CCIS	Night Connection - Flexible - CCIS
Call Transfer - Attendant - CCIS	Outgoing Trunk Queuing - CCIS
Called Station Status Display - CCIS	Paging Access - CCIS
Calling Name Display - CCIS	Restriction from Outgoing Calls - CCIS
Calling Number Display - CCIS	Service Display - CCIS
CCIS Networking via IP	Single-Digit Station Calling - CCIS
Centralized Billing - CCIS	Station-Controlled Conference - CCIS
Centralized Day/Night Mode Change - CCIS	Station-to-Station Calling - CCIS
Centralized E911 - CCIS	Station-to-Station Calling - Operator Assistance - CCIS
Consultation Hold - All Calls - CCIS	Toll Restriction 3/6 Digits - CCIS
Data Line Security - CCIS	Trunk Answer from Any Station - CCIS
Deluxe Traveling Class Mark - CCIS	Trunk-to-Trunk Restriction - CCIS
Dial Access to Attendant - CCIS	Uniform Numbering Plan - CCIS
Digital Display - Station - CCIS	Voice Call - CCIS
Digital Display - Trunk - CCIS	Voice Mail Integration - CCIS
Direct-In Termination - CCIS	Voice Mail Live Record - CCIS
Distinctive Ringing - CCIS	Voice Mail Private Password - CCIS
Do Not Disturb - CCIS	
Dual Hold - CCIS	
Elapsed Time Display - CCIS	
Flexible Numbering of Stations - CCIS	
Hands-Free Answerback - CCIS	
Hot Line - CCIS	
House Phone - CCIS	
Incoming Call Identification - CCIS	
Individual Attendant Access - CCIS	

CCIS Features Descriptions

Attendant Camp-On with Tone Indication – CCIS

This feature permits the Attendant, when the desired station at another switching office is busy, to hold an in-coming call in a special waiting mode. A distinctive Camp-On tone is sent to the busy station when the Attendant sets Camp-On. When that station becomes idle, it is automatically rung and connected to the waiting trunk party.

Attendant Controlled Conference - CCIS

This feature permits an Attendant (2400 IPX) to establish a conference, through CCIS, with up to eight parties of stations and/or trunks (inside and outside parties).

Automatic Recall - CCIS

This service feature works as a time reminder. When an Attendant-handled call through CCIS remains on hold, camped-on, or ringing unanswered for a fixed interval, the Attendant is automatically alerted.

Brokerage - Hot Line – CCIS

This feature provides a ringdown connection between two stations, each using a Multiline Terminal, in different offices in the CCIS network.

Busy Lamp Field (BLF) - CCIS

This feature provides a busy status indication of the predetermined stations within the CCIS network. The visual indication is provided with a red LED associated with each DSS button on the DSS/BLF Console and Multiline Terminal. Pressing the DSS button allows a direct access to the preprogrammed station within the CCIS network.

Busy Verification - CCIS

This feature permits an Attendant, via the Attendant Console on the 2400 IPX or the 2000 IPS, to interrupt a busy station's call at another switching office connected through CCIS.

Call Back - CCIS

This feature provides inter-office Call Back. A station that has dialed a busy station at another office can set Call Back - CCIS by dialing a feature access code. When this feature has been set, the setting station will ring as soon as the busy station becomes available.

Call Forwarding - All Calls - CCIS

This feature permits all calls destined for a particular station to be routed to another station or to an Attendant Console, in another office in the CCIS network, regardless of the status (busy or idle) of the called station. The activation and cancellation of this feature may be accomplished by either the station user or an Attendant.

Call Forwarding - Busy Line – CCIS

This feature permits a call to a busy station to be immediately forwarded to a pre-designated station or to an Attendant Console in another office in the CCIS network.

Call Forwarding – Don't Answer - CCIS

This feature permits a call to an unanswered station to be forwarded to a pre-designated station or to an Attendant Console in another office, when the called station does not answer after a predetermined time period.

Call Forwarding - Intercept - CCIS

This feature allows calls to an inoperative number, through a CCIS trunk, to be intercepted and automatically routed to a recorded announcement informing the caller that an inoperative number was dialed and giving the Listed Directory Number for information.

Call Forwarding - Override - CCIS

This feature allows a target station user (Station A) to call a station (Station B) that has set Call Forwarding - All Calls - CCIS.

Call Processing Indication - CCIS

This feature provides visual indications of all CCIS calls being processed or waiting processing at the Attendant Console.

Call Transfer - All Calls - CCIS

This feature allows a station user to transfer incoming or outgoing Central Office, intra-office and inter-office calls to another station in the CCIS network, without Attendant assistance.

Call Transfer - Attendant - CCIS

This feature permits a station user, while connected to a CCIS network call, to transfer a call to an Attendant Console via the CCIS network.

Called Station Status Display - CCIS

This service feature provides, on the LCD display of the calling Multiline Terminal, a display of the called station status of the remote office within the CCIS network.

Calling Name Display - CCIS

This feature permits the station name of a calling or called party at another switching office, through the CCIS network, to be displayed either on a Multiline Terminal or an Attendant Console.

Calling Number Display - CCIS

This feature permits the number of a calling or called party at another switching office to be displayed either on a Multiline Terminal or an Attendant Console.

CCIS Networking via IP

This feature provides CCIS networks with Voice over IP (VoIP) capabilities to provide a converged infrastructure over corporate Wide Area Networks (WAN). The IP Enabled Dterm can communicate with other IP Enabled Dterm over the CCIS network (IP based) on a peer-to-peer connection basis. The legacy terminals (TDM based terminals) can communicate with other legacy terminals or IP Enabled Dterm terminals over the CCIS network, via IP Packet Assembler/ Disassembler (IP-PAD). Voice compression of G.729a (8 kbps) and G.723.1 (5.3 kbps / 6.3 kbps) is available for those connections. The CCIS Networking via IP provides users with all TDM-based CCIS functionality, such as feature transparency, centralized management, and centralized facilities.

Centralized Billing - CCIS

This feature is used to collect billing information from each office within the network and to direct it to the associated center office. Billing information is then forwarded to the central billing centers via RS232C interfaces.

Centralized Day/Night Mode Change - CCIS

This feature switches the Day/Night mode of a remote office, linked to the main office (2400 IPX) via CCIS. The mode is switched in accordance with the Day/Night mode switching on the Attendant Console at the main office.

Centralized E911 – CCIS

This feature allows the system to transmit a calling party number to the 911 Emergency system over CCIS tandem connection.

Consultation Hold - All Calls - CCIS

This feature permits a station user, within the CCIS network, to hold any incoming or outgoing public network or Tie Line call while originating a call to another station within the CCIS network.

Date Line Security- CCIS

This service feature allows the lines, which are used for data transmission through CCIS to be, protected from interruptions such as Attendant Camp - On, Busy Verification - CCIS, Executive Right of Way, and Attendant Override.

Deluxe Traveling Class Mark - CCIS

This feature provides outgoing call restrictions within the CCIS network. The following three types of restrictions are allowed:

- Deluxe Traveling Class Mark Restriction
- Route Restriction
- Numbering Restriction

Dial Access to Attendant – CCIS

This feature allows a station user to call an Attendant Console by dialing an operator call code through the CCIS network.

Digital Display - Station - CCIS

This service feature provides a display of the station number on the Attendant Console, during an Attendant-to-station connection, within the CCIS network.

Digital Display - Trunk - CCIS

This service feature provides the Attendant with a visual indication of incoming and outgoing trunk calls during an attendant-to-trunk connection within the CCIS network. Trunk Group number and trunk identification code are displayed.

Direct-In Termination - CCIS

This feature automatically routes incoming exchange calls through CCIS to a pre-assigned station in the network, without Attendant assistance.

Distinctive Ringing – CCIS

This feature provides distinctive station ringing patterns for terminated calls, through the CCIS network, so that a station user can distinguish between incoming internal and external calls.

Do Not Disturb - CCIS

This feature allows a station user to establish Do Not Disturb (DND) status on a temporary basis, during which time access to the station from CCIS calls will be denied.

Dual Hold – CCIS

This feature allows two connected Multiline Terminals to be placed on hold simultaneously over the CCIS link. This enables the held parties to answer or originate a call from a secondary extension or the idle primary extension.

Elapsed Time Display – CCIS

This feature provides an LCD, which shows the duration of time that a Multiline Terminal is connected to any trunk through the CCIS network.

Flexible Numbering of Stations – CCIS

This feature allows voice and data station numbers to be assigned to any instrument in the CCIS network, based solely upon numbering plan limitations.

Hands-Free Answerback - CCIS

This feature allows a Multiline Terminal station user to respond to a voice call, through the CCIS network, without lifting the handset.

Hot Line – CCIS

This feature allows two stations, at different nodes in the CCIS network, to be mutually associated on an automatic ringdown basis through the CCIS network.

House Phone – CCIS

This feature allows selected stations to call an Attendant Console, through the CCIS network, simply by going off hook.

Incoming Call Identification - CCIS

This feature allows an Attendant to visually identify the type of service and/or trunk group, which is arriving or waiting to be answered at the Attendant Console through the CCIS network.

Individual Attendant Access – CCIS

This feature permits a station user to call a specific Attendant Console, in the CCIS network, using an individual Attendant Identification Number.

LDN Night Connection – CCIS

This feature routes Listed Directory Number (LDN) calls to a pre-selected station, in the CCIS network, when the Night mode has been entered.

Link Alarm Display - CCIS

This feature provides an indication on external equipment when the CCIS link is connected/disconnected, when the system is initialized, or when the CCH is in make-busy.

Link Reconnect - CCIS

This feature provides the system connected to CCIS network with the capability to release the redundant CCIS link connection and re-connect the link within the system for efficient usage of the CCIS links.

Message Waiting Lamp Setting – Attendant – CCIS

This feature allows an Attendant, in the 2400 IPX, to set or cancel a Message Waiting lamp indication, through the CCIS network, on a station in 2000 IPS.

Message Waiting Lamp Setting - Station – CCIS

This feature allows a station user, in the 2400 IPX, to set or cancel a Message Waiting lamp indication, through the CCIS network, to a station in 2000 IPS with this feature.

Miscellaneous Trunk Access - CCIS

This feature provides access to all types of external and customer-provided equipment/facilities, such as Tie Line and exchange network, along with Dictation, Paging Access - CCIS and Code Calling through the CCIS network.

Miscellaneous Trunk Restriction - CCIS

This feature denies certain stations and dial-repeating tie trunks access to specific trunk groups, such as Tie Line, exchange network, Dictation or Paging Access - CCIS through the CCIS network.

Multiple Call Forwarding - All Calls - CCIS

This feature allows the last hop of a Multiple Call Forwarding - All Calls sequence to be forwarded over a CCIS network to a station in another office.

Multiple Call Forwarding - Busy Line – CCIS

This feature allows the last hop of a Multiple Call Forwarding - Busy Line sequence to be forwarded over a CCIS network to a station in another office.

Multiple Call Forwarding - Don't Answer - CCIS

This feature allows the last hop of a Multiple Call Forwarding - Don't Answer sequence to be forwarded over a CCIS network to a station in another office.

Multiple Console Operation – CCIS

This feature provides console operation where Attendant Consoles are installed in more than one node in the CCIS network.

Network Station Number - CCIS (FCCS)

When 2000 IPS is connected to a 2400 IPX via CCIS link, the Network Station Number can be moved to other office within the network by a simple command operation from the Centralized MAT in the 2400 IPX.

Night Connection - Fixed – CCIS

This feature routes calls normally directed to the Attendant Console to a pre-selected station in another office, through the CCIS network, when the Night mode has been entered.

Night Connection - Flexible – CCIS

This feature provides an inter-office night connection service, via the CCIS network, when the calling station and the night station belong to different offices.

Outgoing Trunk Queuing - CCIS

This feature allows a CCIS network station, upon encountering an all trunk busy signal, to dial a specified access code and enter a first-in, first-out queue. As soon as a CCIS trunk becomes available, stations in the queue will be called back on a first-come, first-served basis.

Paging Access - CCIS

This feature provides dial access to paging equipment from an Attendant Console or a station, through the CCIS network.

Restriction from Outgoing Calls - CCIS

This feature automatically restricts users of pre-selected stations from placing outgoing calls and/or certain miscellaneous trunk calls through CCIS, without Attendant assistance.

Service Display - CCIS

This feature generates LCD displays on the Multiline Terminal corresponding to the various features as they are initiated.

Single-Digit Station Calling - CCIS

This feature allows the assignment of Single-Digit Station numbers.

Station-Controlled Conference - CCIS

This feature allows any station of the 2400 IPX to establish a conference among a maximum of eight parties of stations and/or trunks (inside and outside parties) of the 2000 IPS, through CCIS.

Station-to-Station Calling – CCIS

This feature permits any station user to dial another station directly, through CCIS, without Attendant assistance.

Station-to-Station Calling - Operator Assistance – CCIS

This feature allows a station user to call another station in the CCIS network, with the assistance of an Attendant Console operator.

Toll Restriction - 3/6 Digits - CCIS

This feature allows the system to be programmed to restrict outgoing calls, through CCIS, according to specific Area and/or Office Codes. This restriction is determined on the basis of a three-digit Area Code or six-digit Area and Office Code numbering plan.

Trunk Answer From Any Station - CCIS

This feature allows any station, not restricted from incoming calls, to answer incoming calls when the network is in Night mode.

Trunk-to-Trunk Restriction - CCIS

This feature allows Trunk-to-Trunk tandem restriction by caller's information sent from each office (e.g., caller is a trunk) through the CCIS network.

Uniform Numbering Plan – CCIS

In a CCIS network, a Uniform Numbering Plan enables a station user to call any other station in the network. Two alternative numbering plans are provided. In the first plan, the station user dials any digit station number from three to eight. The location of the office is identified by the first one-, two-, or three-digit of the station number. In the second plan, the station user dials a one-, two- or three-digit office code and any digit station number from two to eight.

Voice Call - CCIS

This feature provides a voice path, through the CCIS network, between a Multiline Terminal in one office and a Multiline Terminal in another office. This path is established from the calling party to the called party's built-in speaker. If the called party's MIC lamp is on, the called party can have a conversation in hands-free.

Voice Mail Integration – CCIS

This feature allows any station user in the CCIS network to utilize the Voice Mail System (VMS) with the Message Center Interface (MCI).

Voice Mail Live Record - CCIS

This feature allows a Multiline Terminal user to record the conversation with another station/trunk within the CCIS network by the Centralized Voice Mail. The conversation in the office without Voice Mail can be recorded to Centralized Voice Mail in other office via CCIS network.

Voice Mail Private Password - CCIS

Voice Mail Password can be prevented from displaying in LCD of Multiline Terminals when connected to the voice mail system via CCIS.

ISDN Feature List

Addressing
Advice of Charge - Display
Call-By-Call Service Selection
Called Party Recognition Service (Direct-In Termination (DIT))
Channel Negotiation
Connected Line Identification Presentation (COLP) / Connected Line Identification Restriction (COLR)
CPN To Network - Present
CPN To Terminating User - Display
DID Addressing
DID and DOD Addressing
Event Based CCIS
Incomplete Number Handling
ISDN Terminal
MEGACOM® Access/WATS
MEGACOM® 800 Service/800 WATS/Ultra WATS
MULTIQUEST®/900 Service
Overlap Receiving
Overlap Sending
Subaddress - Present
Trunk Provisioning Service Selection

ISDN Feature Descriptions

Addressing

The *UNIVERGE* NEAX 2000 IPS supports E.164 numbering plan as Type of Number (TON) / Numbering Plan Identifier (NPI) combination in Calling and Called Party Number Information Element for ISDN calls.

Advice of Charge - Display

When an outgoing connection is made on the ISDN trunk, this feature provides the call originator with the charge information form the network. The information will be displayed on the calling Dterm LCD for six seconds after the station has been released, and be output to the SMDR port.

Call-By-Call Service Selection

Services can be selected on a call-by-call basis to all channels of a single PRI interface according to applications. Services may be used on any available channel, unlike Trunk Provisioning Service, in which services are assigned to specific channels.

Called Party Recognition Service (Direct-In Termination (DIT))

This feature provides an incoming Direct-In Termination (DIT) call via an ISDN trunk to be connected to a predetermined station. This application can be used for a station or modem.

Channel Negotiation

When collisions of data occur between an outgoing call to ISDN and an incoming call from ISDN, the B channel that 2000 IPS specified will be busy on the network and the network will specify another B channel. This feature provides negotiation for the selection of a B-channel between the 2000 IPS and the network, when the SETUP message is delivered.

Connected Line Identification Presentation (COLP) / Connected Line Identification Restriction (COLR)

Connected Line Identification Presentation (COLP) provides the 2000 IPS to send the ISDN number of the connected party in the 2000 IPS, to the calling party. Connected Line Identification Restriction (COLR) provides the 2000 IPS to restrict presentation of the connected party's ISDN number in the 2000 IPS, to the calling party.

CPN To Network–Present

This feature allows the ISDN network to be informed of the Calling Party Number (CPN) when a call originates from a terminal connected to the *UNIVERGE* NEAX 2000 IPS.

CPN To Terminating User–Display

This feature provides a visual display of the originating station's number and sub-address information on a Multiline Terminal for incoming ISDN calls. This provides the terminal user with a quick and accurate way to identify the originating station's number (Calling Number).

DID Addressing

This feature allows incoming ISDN-PRI calls to terminate to stations, Attendant Console, Automated Attendant, etc., based on the Called Party number. Direct Inward Dial trunks will be terminated to programmed destinations without Attendant assistance.

DID and DOD Addressing

This feature allows the system to use DID and DOD on the same B-Channels. Trunk Provisioning Service election is not required. (B-Channels can be used for DID and DOD without separating the trunk routes.)

Event-Based CCIS

This feature allows a PBX customer who does not have a tie line (or when a customer cannot use the tie line due to busy or fault of the line), to use the various CCIS feature by using the ISDN line as a CCIS virtual tie line, between the 2400 IPX to 2000 IPS connection or the 2000 IPS to IPS connection.

Incomplete Number Handling

When a DID number incoming from ISDN is incomplete, the call can be transferred to a predetermined station or attendant console.

ISDN Terminal

This feature provides the system with an ISDN Terminal or Terminal Adapter (TA). ISDN Terminal to ISDN Terminal, ISDN Terminal to ISDN Trunk, ISDN Trunk to ISDN Terminal, ISDN Terminal to Single Line Telephone, ISDN Terminal to Multiline Terminal, and ISDN Terminal to PS connections are available.

MEGACOM® Access/WATS

AT&T's MEGACOM® (WATS) network, as well as WATS from other carriers, can be used.

MEGACOM® 800 Service/800 WATS Ultra WATS

AT&T's MEGACOM® 800 (Inward WATS) network, as well as 800 WATS provided by other carriers, can be used.

MULTIQUEST® /900 Service

AT&T's MultiQuest® service can be used. (It is a "900"-type service.) Also, 900 service provided by other carriers can be used.

Overlap Receiving

Overlap Receiving is available for an incoming call to the 2000 IPS from ISDN network. Overlap Receiving is a procedure, used in call establishment of an incoming call, to enable the network to send called party number digits to the user in successive messages, as and when they are made available from the remote network.

Overlap Sending

Overlap Sending is available for an outgoing call from the 2000 IPS to ISDN network. Overlap Sending is a procedure, used in call establishment of an outgoing call, to enable the user to send called party number digits to the network in successive messages.

Sub-Address–Present

This feature allows a Primary Rate Interface ISDN trunk to transfer the Called Party Sub-address information to a destination ISDN station when the call is originated by the system. Dialing the called party station number and sub-address is required.

Trunk Provisioning Service Selection

Each channel of a PRI interface can be dedicated to a particular service. Services are designated to specific channels; once designated, a channel can be used only for that service.

Q-SIG Feature List

Calling/Connected Line ID Presentation (CLIP/COLP)/ Calling/Connected Name ID Presentation (CNIP/CONP)
CCIS Tandem Call-Calling Party Number (CPN) Delivery to ISDN & Q-SIG Networks
Q-SIG Circuit Switched Basic Call - ETSI Version

Q-SIG Feature Descriptions

Calling/Connected Line ID Presentation (CLIP/COLP)/ Calling/Connected Name ID Presentation (CNIP/CONP)

In between Q-SIG networks, this feature allows calling or called party information, to be displayed on the Dterm LCD. CLIP/COLP conforms to ETS 300 173 and IS-11572 network. CNIP/CONP conforms to ETS 300 238 and IS-11572 network.

- Calling Line Identification Presentation (CLIP): Calling party number (ID) is displayed on the called party's Dterm LCD.
- Connected Line ID Presentation (COLP): Called party number (ID) is displayed on the calling party's Dterm LCD.
- Calling Name Identification Presentation (CNIP): Calling party information (Name ID) is displayed on the called party's Dterm LCD.
- Connected Name Identification Presentation (CONP): Called party information (Name ID) is displayed on the calling party's Dterm LCD.

This feature is also available when networking with a CCIS interface.

CCIS Tandem Call-Calling Party Number (CPN) Delivery to ISDN & Q-SIG Networks

This feature is provided for the call from tie line (CCIS line or Q-SIG) or a station, is sent to the public ISDN network (AT&T, NT, NI-2) or Q-SIG network.

Q-SIG Circuit Switched Basic Call - ETSI Version

This feature enables NEC's PBX to connect to NEC's PBX or other manufacturer by using Layer 3 protocol for the signaling for the support of circuit mode bearer services at the Q reference point.
Q-SIG conforms to ETS 300 172.

Wireless Feature List

Analog PBX Interface
Announcement - PS No Answer / Announcement - PS Out of Zone
Automatic Release - Out of Zone Calls
Call Forwarding - Not Available
Call Forwarding - PS Out of Zone
CCIS Interface
Calling Name Display - PS
Calling Number Display - PS
DTMF Signal Sender
Group Call - Automatic Conference (6/10-Party)
Group Call - 2 Way Calling
Handover
Individual PS Calling
Last Number Redial - PS
Multi-Line Operation - PS
Multi-Site Roaming
UNIVERGE NEAX 2000 IPS Wired for Wireless
Number Sharing
Out of Zone Indication
Overlap Dialing
Preset Dialing
PS Authorization
PS Location Indication
PS Location Registration
Q. 931a Roaming over IP Trunk
Radio Channel Changeover
Short Message Notification (OAI)
Soft Phone Wireless Handset
Speech Encryption
Speed Dial - PS
Station Hunting - Not Available
Voice Mail Indication
WLAN Handset Registration/Service

Table of Features by Configuration Type

Feature	Adjunct Type		Integrated Type
	Analog PBX Interface	CCIS Interface	*UNIVERGE* NEAX 2000 IPS Wired for Wireless
Analog PBX Interface	X	–	–
Announcement - PS No Answer	X	X	X
Announcement - PS Out of Zone	X	X	X
Automatic Release - Out of Zone Calls	X	X	X
Call Forwarding - Not Available	–	–	X
Call Forwarding - PS Out of Zone	–	–	X
CCIS Interface	–	X	–
Calling Name Display - PS	–	X	X
Calling Number Display - PS	–	X	X
DTMF Signal Sender	X	X	X
Group Call - Automatic Conference(6/10-Party)	–	–	X
Group Calling - 2 Way Calling	–	–	X
Handover	X	X	X
Individual PS Calling	X	X	X
Last Number Redial - PS	X	X	X
Multi-Line Operation - PS	X	X	X
Multi-Site Roaming	–	X	X
UNIVERGE NEAX2000 IPS Wired for Wireless	–	–	X
Number Sharing	–	–	X
Out of Zone Indication	X	X	X
Overlap Dialing	X	X	X
Preset Dialing	X	X	X
PS Authorization	X	X	X
PS Location Indication	–	–	X
PS Location Registration	X	X	X
Q. 931a Roaming over IP Trunk	–	–	X
Radio Channel Changeover	X	X	X
Short Message Notification (OAI)	–	–	X
Soft Phone Wireless Handset	–	–	X
Speech Encryption	X	X	X
Speed Dial - PS	X	X	X
Station Hunting - Not Available	–	–	X
Voice Mail Indication	–	–	X
WLAN Handset Registration/Service	–	X	X

X = available — = not available

Wireless Feature Descriptions

Analog PBX Interface

This feature allows the WCS to be integrated with many types of PBXs and to activate various PBX services by sending hook flash signal and DTMF tone.

Announcement - PS No Answer / Announcement - PS Out of Zone

This feature allows calls to a Dterm PS III which cannot be paged in a predetermined period of time to be routed to the announcement notifying the calling party that the Dterm PS III cannot answer. Announcements can be divided between Announcement-PS No Answer and Announcement-PS Out of Zone depending on the Dterm PS III condition.

Automatic Release - Out of Zone Calls

When a Dterm PS III user engaged in a call moves out of the service area or enters a radio zone where all the channels are busy and the communication becomes unavailable, the call will be disconnected automatically, and the other party receives reorder tone.

Call Forwarding-Not Available

When a Dterm PS III is powered off or out of zone, a call directed to the Dterm PS III is forwarded to a VMS, and a voice mail message can be recorded to the VMS and checked from the Dterm PS III. Also the VMS can page the Dterm PS III automatically after the voice mail message is recorded.

Call Forwarding - PS Out of Zone

This feature allows calls directed to a Dterm PS III, which is power off or out of zone to be routed to an another station, an Attendant, an outside number or voice mail equipment. The call forwarding destination can be set/ cancelled from the Dterm PS III station or MAT/CAT.

CCIS Interface

This feature allows the WCS to be integrated with NEC PBXs with CCIS interface.

Calling Name Display - PS

Without answering incoming calls or hold calls terminates to the Dterm PS III, the calling party's name can be confirmed by the indications on the LCD.

Calling Number Display - PS

This feature provides a display on the LCD of a Dterm PS III receiving a call, indicating the station number.

DTMF Signal Sender

This feature allows a Dterm PS III user to send the DTMF signal (tone) to the called party (terminal, voice mail system, etc.) while engaged in communication.

Group Call-Automatic Conference (6/10-Party)

This feature permits a Dterm PS III user, Multiline Terminal user or Single Line Telephone user within the system to establish a conference among as many as six or ten parties. From a Dterm PS II/D term /Single Line Telephone, a maximum of 9 PSs can be paged simultaneously except the conference leader. The PSs are assigned to the simultaneous paging groups as participants by the system data beforehand.

Group Call – 2Way Calling

This feature permits a Dterm PS III, Multiline Terminal and Single Line Telephone to page a maximum of fifteen parties simultaneously, except the group call leader. After one of paged parties answers, the paging becomes the 2 Way Calling between the leader and the first answered party, and paging other parties stops automatically. The Dterm PS III is assigned through system data as participants in the simultaneous paging groups.

Handover

When the signal transmission quality becomes inferior, a Dterm PS III re-originates a call automatically and seizes another radio channel, making the WCS handover the call to another zone transceiver to maintain the speech quality.

Individual PS Calling

This feature allows the calling party to page the individual Dterm PS III.

Last Number Redial – PS

This feature enables a Dterm PS III to store the numbers dialed in the previous five calls including the last number dialed. The stack dial numbers are sequentially displayed on the LCD, allowing the station user to make an outgoing call by selecting the desired dialed number from the display.

Multi-Line Operation – PS

Dterm PS III equipped with two line keys, L1 key and L2 key, and different station numbers can be assigned to each of two lines. The number assigned to L1 key of the PS is called My Line and the other number assigned to L2 key is called Sub Line. My Line and Sub Line of a Dterm PS III can be assigned to appear on the Flexible Line Keys of a Multiline Terminal and the Multiline Terminal can share the Dterm PS III lines.

Multi-Site Roaming

Dterm PS II user can originate or receive a call in any place of a network provided by multiple PBXs which communicate through JT-Q931-a.

***UNIVERGE* NEAX 2000 IPS Wired for Wireless**

This feature allows the *UNIVERGE* NEAX 2000 IPS to have the Wireless PBX feature. The IPS provides several integrated features adding on the adjunctive configuration.

Number Sharing

This feature allows the Multiline Terminal user to have a Dterm PS III as Sub Station and to get service with one telephone number. In case that one user has both a Multiline Terminal and Dterm PS III, with this feature used the user is not required to have two separate telephone numbers.
-When a user is at his desk, a call is terminated to his Dterm.
-When a user leaves their desk with Dterm PS III, a call is automatically terminated to the Dterm PS III.
In this feature, the Multiline Terminal and Dterm PS III are referred as Main Station and Sub Station, respectively.

Out of Zone Indication

When a Dterm PS III user moves out of the service area and the electric field strength becomes weak, this feature notifies the user with a warning tone and an LCD display.

Overlap Dialing

This feature allows a Dterm PS III user to receive dial tone and dial the desired number to originate a call.

Preset Dialing

This feature allows a Dterm PS III user to confirm the number to be dialed before originating a call.

PS Authorization

This feature is to confirm the identity of a Dterm PS III to avoid an unauthorized Dterm PS III from accessing the system.

PS Location Indication

When a Dterm PS III calls a Multiline Terminal/Attendant Console or vice versa, this feature allows to indicate the location name of the ZT II that is connected to the Dterm PS III on the LCD of the Multiline terminal.

PS Location Registration

This feature allows the IPS-WCS to supervise the location of each Dterm PS III, upon receiving the location registration request, to allow call termination.

Q931a Roaming Over IP Trunk

Dterm PS III user can originate or receive a call in any place of a network provided by multiple PBXs, which are interfaced by JT-Q931-a protocol over IP Trunks.

Radio Channel Changeover

This feature is to supervise and changeover the channel automatically to avoid the interference and to maintain the speech quality.

Short Message Notification (OAI)

This feature enables a Short Text Message (STM), once arrived at a mail box of the STM Server (external equipment), to be automatically distributed to the addressee Dterm PS III via Open Application Interface (OAI). This feature also provides “TM Full” notification on the display (LCD) of the address D term PSII that is busy. If the Dterm PS III is out of zone, the feature makes retransmission of the STM after the PS returns to the zone.

Soft Phone Wireless Handset

This feature allows a Soft Phone (Dterm SP30) user to use a PS terminal as the handset of the Soft Phone. The user can select a handset (Soft Phone Handset or PS) by pressing the Handset Switch key on the Soft Phone.

Speech Encryption

This feature protects a call from being tapped.

Speed Dial-PS

This feature allows a Dterm PS III user to dial the certain frequently called numbers using two-digit abbreviated call codes.

Station Hunting - Not Available

This feature allows a call placed to a Dterm PS III station, which is out of zone or power off to be forwarded to an idle station in a hunt group. Two Station Hunting arrangements are available. Station Hunting - Circular processes the call regardless of which station in the hunt group is called. Station Hunting - Terminal initiates a hunt only when the pilot number of a hunt group is called.

Voice Mail Indication

When a message is mailed in the Dterm PS III, an indication of the envelope icon is displayed in the LCD of the Dterm PS III.

WLAN Handset Registration/Service

This feature allows the 2000 IPS to accommodate the WLAN Handset as a station. A terminal is connected to the 2000 IPS through Access Point and Gateway and operates under the control of the virtual CS on the Gateway. Station-to-Station calls between WLAN stations and calls between WLAN stations and existing stations/trunks are available.

Chapter 10 System Input/Output (I/O) Interfaces

Station Message Detail Recording (SMDR)

This feature provides a call record for outgoing Station-to-Trunk calls, incoming Trunk-to-Station calls (including Data Call), and tandem calls. This facilitates cost control by identifying trunk use and misuse by individual stations. Station Message Detail Recording (SMDR) enables call billing to customers and clients, and provides a means for checking local telephone bills. The PBX provides three kinds of Station Message Detail Recording (SMDR) formats; 2400 IMS Format, 2400 IMS Extended Format and 1400 IMS Format.

2400 and 2400 Extended Format

The following call information is provided to the SMDR terminal upon completion of each outgoing or incoming C.O. call.

- Type of Record
- Trunk Number (000-255)
- Trunk Route Number (00-63)
- Calling/Called Party Information
  - Calling/Called Party I.D.
  - Tenant Number (01-63)
  - Station Number (Max. 5 digits)
  - Attendant Number (00-07)
  - Trunk Route (00-63) + Trunk Number (000-255)
- Time of Start Conversation-Month, Day, Hour, Minute, Second
- Time of Call Completion-Month, Day, Hour, Minute, Second
- Account Code (Max. 10 digits)
- Condition Code
- Route Advance Information
- Called Number (Max. 24 digits)*
- Call Metering Information
- Calling Party Office Number (Max. 4 digits)
- Billing Office Number (Max. 4 digits)
- ANI Information (Former: Max. 16 digits/Extended: Max. 32 digits)*
- Authorization Code (Max. 10 digits)*
- Call Start Year
- Call End Year
- Condition Code for Advice of Charge (AOC)*
- Advice of Charge (AOC)*

Note: *The information marked by "*" is only for outgoing C.O. calls.*
*The information marked by "**" is only for incoming calls.*

1400 IMS Format

1400 IMS Format is generally used to print call records on a serial printer and provides an easy to read format for printouts of outgoing / incoming call activity.

- Each record is given a serial number from 0000 to 9999 to identify each call.
- Calling Station Number
- Called Number (26 digits maximum)
- Route Number and Trunk Number
- Start of Call Time (from month down to second)
- Duration of Call (hours, minutes, seconds, rounded to nearest .5 sec.)
- Forced Account / Authorization Code
- Account Code
- Attendant Handled
- Trunk Route Overflow (Route Advance)
- CPN/ANI up to (16 digits)
- Six options for 80/136 character printers

SMDR Terminal Interface

A Personal Computer or a Host Computer is used as the SMDR terminal for receiving and processing the call information that arrives at the MP. The transmission interface specification by which the AP or MP sends out call information to the SMDR terminal are as follows:

- SMDR with AP00/MP built-in SMDR on RS-232C

ITEM	Specifications	
	SMDR w/AP00	MP Built-in SMDR on RS-232C
Physical Interface	RS-232C	RS-232C
Synchronization	Asynchronous	Asynchronous
Protocol	Non protocol (Free Wheel)	Non protocol (Free Wheel)
Transmission Speed	1200/2400/4800/9600 bps (for PN-AP00-B with MRC-C program) **Note 1** 300/1200/2400/4800 /9600/19200 bps (for PN-AP00-B with MRC-F program)	1200/2400/4800/9600/19200 bps
Stop Bit	1/1.5/2 bits	1/2 bits
I/O Port	No. 0-3 port of AP00-B card	RS port of MP card
Terminal Busy Detecting Method	Data Carrier Detect (DCD) signal ON/OFF (terminal ready/busy)	Data Carrier Detect (DCD) signal ON/OFF (terminal ready/busy)

Note 1: *For the port 1 and 3 of AP00-B card with MRC-C program, data speed cannot be set to 9600 bps.*

Note 2: *Upon confirming that the status of the DCD signal from the SMDR terminal is ON, the system sends out call information to the SMDR terminal. When the status of the DCD signal is OFF, the system does not send out call information but temporarily stores the information until the SMDR terminal becomes ready to receive call information, in other words, until the status of the DCD signal changes to ON. If the status of the DCD signal has changed from ON to OFF while transmission of specific call information is in progress, the next call information is not sent out but stored into the system temporarily. The same applies to CTS and DSR signals.*

Note 3: *For output to MP built-in SMDR, one message is sent at intervals of one second unidirectional.*

- MP built-in SMDR on IP

ITEM	Specifications
Physical Layer	Ethernet
Connection Layer	The Ethernet packet format complies with the DIX standard
TCP/IP Protocol	ARP, IP, ICMP, UDP, TCP
Socket Interface	Complies with 4.3 BSD socket interface
Transport Protocol	TCP stream type protocol
Application Port Number	60010 (fixed)
Number of Connections	1
Client/Server	Client: SMDR terminal Server: PBX
Transmission Code	7-bit ASCII code
Quasi-normal Restriction Condition	1. When connection is closed 2. Status monitoring text

Note 1: *The MP card in main site communicates with the SMDR terminal. Therefore, in the communication settings at SMDR terminal side, set the IP address to connected to the address specified by office data (CM0B Y=00>00 or CM0B Y=02>03), and application port number shown in the above table.*

Note 2: *One message is sent each time a request is received from SMDR terminal.*

Combinations between SMDR and SMDR Output Format

SMDR	2400 IMS Format	Extended 2400 IMS Format	1400 IMS Format
SMDR with AP00 (PN-AP00-B with MRC-C program)	X	—	X
SMDR with AP00 (PN-AP00-B with MRC-F program)	X	X	—
MP built-in SMDR on RS-232C	X	X	—
MP built-in SMDR on IP	X	X	—

X: Available —: Not available

Combinations available at the same time

SMDR	Pattern A	Pattern B	Pattern C	Pattern D	Pattern E
SMDR with AP00 (PN-AP00-B with MRC-C program)	—	—	—	—	X
SMDR with AP00 (PN-AP00-B with MRC-F program)	—	X	—	X	—
MP built-in SMDR on RS-232C	X	X	—	—	—
MP built-in SMDR on IP	—	—	X	X	—

X: Available —: Not available

Combinations for Center Office of Centralized Billing

SMDR	Former 2400 IMS Format	Extended 2400 IMS Format
SMDR with AP00 (PN-AP00-B with MRC-C program)	X	—
SMDR with AP00 (PN-AP00-B with MRC-F program)	X	X
MP built-in SMDR on RS-232C	—	—
MP built-in SMDR on IP	X	X

X: Available —: Not available
Note: *Only one output can be used at the center office.*

Combinations for Local Office of Centralized Billing

SMDR	Former 2400 IMS Format	Extended 2400 IMS Format
SMDR with AP00 (PN-AP00-B with MRC-C program)	X	—
SMDR with AP00 (PN-AP00-B with MRC-F program)	X	—
MP built-in SMDR on RS-232C	X	X
MP built-in SMDR on IP	—	—

X: Available —: Not available
Note: *Only one output can be used at the local office.*

Service Conditions

- SMDR can be programmed to record all outgoing calls or toll calls only, depending on the customers' requirements.
- When customer provided computer equipment is connected using the RS-232C interface, SMDR information will be transmitted directly to the equipment as each call record is completed.
- If the outgoing call is directed to a trunk, which does not supply answer supervision, SMDR will start recording the call approximately 10 seconds after the last digit has been dialed.
- Supervision of the status of the external RS-232C terminal is not supplied.
- There are two kinds of SMDR features as follows:

  a. Built-in SMDR
  The MP card has two RS-232C ports for SMDR. This card provides memory for a maximum of 1024 calls, and provides a record of a maximum of 255 trunk calls simultaneously for a maximum of 6-digit extension number.

  Note: *In case of built-in SMDR, if the station number is a 7- or 8- digit number, the upper one or two digits are sent as Calling office Number and the last six digits are sent as Calling Number in the call record format.*

b. SMDR with AP00
- AP00 without expansion memory (standard): up to 1600 calls (if central office feature is used, up to 800 calls)
- AP00 with expansion memory (optional): up to 27000 calls (if central office feature is used, up to 26000 calls)
- When a call is completed, the record is sent to the output device and is removed from memory.
- If the amount of calls exceed the SMDR memory, those overflow calls will not be recorded.

- The built-in SMDR cannot be provided when AP00B with MRC-C program is mounted.
- Account Codes, Forced Account Codes, Authorization Codes and DISA Codes on Tandem connections (SMDR with AP00 only) are reported in the applicable call record.
- SMDR with AP00 provides a record of incoming Tie Line tandem calls where another trunk is dial accessed.
- For details of SMDR format, data stream, memory buffer, etc., refer to the Installation Procedure Manual/Feature Programming Manual.
- Up to 16 digits of the calling subscriber's number for Automatic Number Identification (ANI) and CPN (ISDN) can be recorded in the SMDR by system programming.

Data Transmission Sequence for Ethernet Interface

Establishing a connection, outputting billing Information, and releasing a connection are performed between the PBX and the SMDR terminal in the following timings.
- Timing to Establish a Connection
  - A connection is established when a connection request is received from the SMDR terminal.
- Timing to Output Billing Information
  - Billing information is output when a polling request is received from the SMDR terminal connected to LAN.
- Timing to Release the Connection
  - The connection is released when the SMDR terminal does not receive data from the PBX in a predetermined time interval. Also the connection is released when the PBX receives connection release text from the SMDR terminal or does not receive data from the SMDR terminal within a given time interval.

System Capacity

- MP Built-in SMDR

Number of Simultaneous Trunk Calls	Maximum No. Call Records
255	1024

- SMDR with AP00

	When not a center office of Centralized Billing-CCIS	When center office of Centralized Billing-CCIS
No EXPMEM card on AP00	1600	800
EXPMEM card on AP00	27000	26000

Hardware Required for SMDR

Description	Qty	Remarks
SPN-AP00B MRC-C (AP)	1 per system	Control card for SMDR/PMS/H/M Printer/CCIS Centralized SMDR **Note**
SPN-AP00B MRC-F (AP)	1 per system	Control card for SMDR in 2400 &2400 Extended format only, Centralized SMDR **Note**
SPN-CP24-C MP (UA)	1 per system	MP
PZ-M606-A (ETHER)	1 per system	MP Mounted card One per MP
RS RVS-4S CA-C	1 per SMDR, PMS	
RS NORM-4S CA-A	1 per SMDR, PMS	
RS PRT-15S CA-A	1 per Printer	
SMDR Terminal	1 per system	
Printer	as Required	
Modem	as required	

Note: *Only one AP00 can be mounted at on time in the system.*

Message Center Interface (MCI)

This feature provides an interface with a customer supplied Voice Mail System (VMS) which can send Message Waiting lamp control data to the system.

The Message Center Interface (MCI) can provide the following operations:

- When terminating the call to the VMS, the system sends call connection status information to the VMS through the MCI.
- The VMS sends the Message Waiting Lamp on data to the MCI.
- The system, upon receiving this control data from the MCI, illuminates the Message Waiting lamp of the corresponding station.
- The VMS, upon receiving retrieved message information, will send the Message Waiting lamp control data requesting the system to extinguish the Message Waiting lamp of the corresponding station.

Service Conditions

- The Voice Mail System (VMS) is interfaced to the system through the 4LCD/8LC card. (The 4LCD/8LC provides disconnect supervision in the form of a momentary loop open.)
- The UCD or Station Hunting feature is usually provided with the VMS station.
- One RS-232C port on the MP card or the Application Processor (AP00) is required to make a data link with a customer supplied VMS.
- Messages can be retrieved from any Multiline Terminal, DTMF telephone, or the Attendant Console, but not from DP telephones.
- The MCI is available to a direct call or a forwarded call from a station/trunk/Attendant to the VMS.

- Stations can set Call Forwarding or Split Call Forwarding - All Calls, No Answer, and Busy Line to the VMS. The system sends out incoming call information to the VMS. A call to a station that has Call For-warding set to the VMS is automatically answered by the VMS.
- The MCI can control the LCD display of a Multiline Terminal for “MESSAGE” Indication. The number of messages is not displayed.
- When the Message Waiting lamp control is activated with the MCI, the lamp control from the following equipment will not be provided:
  - From the Property Management System (PMS)
  - From the station (by dialing the access code)
  - From the Direct Station Selection (DSS) Console
  - From the Hotel/Motel Attendant Console
  - From the Hotel/Motel Front Desk Instrument
  - From the Attendant Console
- Only one system should be programmed (via system programming) to control Message Waiting lamps through the CCIS network.
- The system controls Message Waiting lamps normally when the time interval between messages is a minimum of 350 msec. or more.
- When the VMS interface line does not answer, all of the messages are sent out from the I/O port of the MP/AP00.
- If the VMS is not ready for information receiving (Busy Status), the AP00 can temporarily store up to 16 call records in its internal memory. If the maximum of 16 call records is stored and a 17th is generated, the system will write over the oldest stored record. When the RS port on the MP card is used for the data link to the VMS, the MP can store up to 15 call records. If a 16th call record is generated when the MP stores 15 call records, the system will write over the oldest stored record.
- The Voice Mail Integration (In band) feature can be combined with voice mail through the MCI in the system. The Voice Mail Integration (In band) feature and MCI feature can coexist in one system and either can be selected per VMS (VMS station number) by system programming.
- When terminating a call with the ANI information to the VMS through the MCI, the system can send the ANI information to the VMS, if required. This is not available through CCIS interface.

Hardware Required for MCI

Description	Qty	Remarks
SPN-CP24-C MP (UA)	1 per system	MP
SPN-AP00B MRC-C (AP)	1 per system	Control card for MCI **Note**
SPN-AP00B MRC-F (AP)	1 per system	Control card for MCI **Note**
RS RVS-4S CA-C	1 per VMS	AP00-VMS, 13.1ft (4.0m)
RS NORM-4S CA-A	1 per VMS	AP00-MODEM, 13.1ft (4.0m)
PN-8LCAA	As required	Interface for VMS (with MW lamps)
VMS	1 per system	
MODEM	As required	

Note: *Only one AP00 can be mounted at on time in the system.*

Property Management System (PMS)

The system provides a data interface to a locally provided Property Management System (PMS). This enables communication between the system and the PMS in order to provide computer control of Hotel/Motel features.

The available PMS feature codes:

Feature	Feature Code	Type of Information	PMS	
			AP00	IP
Maid Status	11	Maid status set up by guest room telephones	X	X
	12	Maid status set up by the Front Desk Terminal		
Message Waiting	13	Message waiting lamp status changes	X	X
Station Message Detail	14	Local/toll call details on completion of calls	X	X
Restriction Control	15	Changes in telephone calling restrictions placed on room station numbers	X	—
Check In/Check Out	16	Room check in/check out status	X	X
Room Data Image	17	Complete status information for a room	X	X
Wake Up	19	Wake up status	X	X
Room Change/Room Swap/Room Copy	20	Room change/room swap status	X	X
Room Occupancy/ Room Data Change	21	Room Occupancy/Room Data Changes	X	—
Data Connection Maintenance	50	Batch processing status	—	X
Message Waiting	53	Controlling message waiting lamps	—	X
Station Message Detail	54	Station message detail information for each call	—	X
Check In/Check Out/Room Change	56	Setting check in/check out and room change	—	X
Room Recovery	57	Room information at system recovery	—	X
Direct Data Entry	59	Data entered from a guest room station	X	X
Room Data Change	61	Room data change information	—	X
Administrative Station Name	65	Setting administrative station name	—	X
Status Inquiry	70	Data link maintenance	X	X

X: Available —: Not available

PMS Interface

- PMS with AP00-B on an RS-232C Interface

Item	Description
Data Rate	1200, 2400, 4800, 9600 bps, asynchronous, software selectable **Note 1**
Operating Mode	Half Duplex
Electrical Interface Characteristic	EIA RS-232C electrical standard interface
Signal Form	EIA RS-404
Interface Distance	Max. 15 m (49.2 ft.) between PBX and PMS (without modem) **Note 2**
Word Framing	10 bits (1 start, 7 data, 1 parity, 1 stop) or 11 bits (1 start, 7 data, 1 parity,2 stop)
Parity VRC	No parity, even parity, odd parity; selected by PBX system data
Parity LRC	Exclusive OR of message text
Frame Contents	US ASCII 7-bit codes
Control	Contention
Priority Sequence	Primary office: PBX Secondary office: PMS

Note 1: *For the Port 1 and Port 3, data speed 9600 bps cannot be used.*
Note 2: *When modems are used, full duplex asynchronous type modems are required.*

- Built in PMS over IP

Item	Description
Physical layer	Ethernet
Connection Layer	The Ethernet packet format complies with the DIX standard
TCP/IP core protocol	ARP, IP, ICMP, UDP, and TCP
Socket interface	Complies with the 4.3BSD socket interface
Transport Protocol	TCP stream-type protocol
Application Port Number	60050 (Fixed)
Number of Connections	1 connection
Client/Server	Server: PBX Client: PMS
Frame Contents	US ASCII 7-bit codes
Pseudo Normal Restriction Condition	1. At connection release 2. Status monitoring text

Service Conditions

1. Only one Interface Port can be provided for PMS.

2. The system sends information relating to the following features to the PMS upon request from the PMS:
   - Do Not Disturb
   - Room Cut-off

3. The *UNIVERGE* NEAX 2000 IPS system can support PMS Model-60 via AP00-B or PMS Model 90/120 via TCP/IP connection.

4. The AP00-B with MRC-F program is used for Executed task Print out Request for MP Built-in PMS.

Hardware Required for PMS

Description	Qty	Remarks
SPN-CP24-C MP (UA)	1 per system	MP
SPN-AP00B MRC-C (AP)	1 per system	Control card for PMS **Note**
SPN-AP00B MRC-F (AP)	1 per system	Executed Task Print Out for MP PMS **Note**
PZ-M606-A (ETHER)	1 per system	MP Mounted card One per MP
RS RVS-4S CA-C	1 per SMDR/PMS	AP00-VMS, 13.1ft (4.0m)
RS NORM-4S CA-A	1 per SMDR/PMS	AP00-MODEM, 13.1ft (4.0m)
RS PRT-4S CA-A	1 per Printer	AP00 Printer
PMS Terminal Serial	1 per system	Model 60 only
PMS Terminal over IP	1 per system	Model 90/Model 120
SMDR Terminal	1 per system	
Printer	As required	
MODEM	As required	

Note: *Only one AP00 at a time can be mounted in the system.*

Chapter 11 Open Application Interface

System Outline

The *UNIVERGE* NEAX 2000 IPS is equipped with an Open Application Interface (OAI) to provide user applications through an external processor, which transmits and receives the control signals via Ethernet interface on the MP. The interface is supports TCP/IP protocol.

Open*WorX*: Attendant Statistics

OpenWorX Statistics operates in conjunction with the Business Attendant System in order to provide near real time statistics about the operators and the queues. A variety of information is gathered and stored in a database that allows for a multitude of differing reports. With the power of Crystal Reports® the user can then view the data with a web interface allowing for HTML, PDF, and RPT formats. An additional purchase of Crystal Reports Viewer® is required for the RPT format. Below is a list of the types of reports the user can run.

Attendant Reports

- Attendant Call Summary
- Attendant Answered Calls
- Attendant Outbound Calls
- Attendant Interval Summary
- Attendant Inbound Interval
- Attendant Outbound Interval
- Total Attendant Summary
- Attendant Login/Logout Detail

Queue Reports

- Queue Summary
- Queue Abandon Calls
- Queue Answered Calls
- Queue Interval
- Total Queue Summary

Open*WorX*: Business Attendant System (BAS)

NEC's Business Attendant System (BAS) is a powerful client/server PC-based tool for full-time and part-time *UNIVERGE* NEAX 2000 IPS system answering position. The BAS consists of an interactive client side application running on Windows 2000 or Windows XP operating system that connects to the BAS server via the corporate LAN. The BAS server software is used for *UNIVERGE* NEAX 2000 IPS connectivity and runs on a Microsoft Windows 2000 Server or Windows 2003 Server.

With a dynamic interactive directory and an Open Application Interface (OAI) driven call queue, the operators are able to process a high call volume with minimal effort and time expenditure in today's sophisticated business environment. For example, an operator can answer an incoming call from one of many queues; find out to whom the caller wishes to speak, begin typing that name to select it in the directory, and once selected, hit the <Enter> key or click the entry to transfer the call. Additionally the caller could be parked, put on hold, or sent to voice mail. Several operators can monitor the same queues and share the same directory for call operation.

BAS: Benefits

- Centralized company database that can link and replicate with other databases, including AIM*WorX*.
- Greeting playback and call recording.
- PC-based call processing.
- Meet-me paging.
- Unlimited, global parked calls.
- Different greetings/displays available for DIDs.
- Single key call processing.
- Directory dialing for accuracy.
- Look ahead status from the directory.
- Sort and searchable directory by key fields.
- Configurable screens for *each* operator.
- Queue types allowing flexibility to answer certain call types.
- Configurable screen pop information.
- MAPI compliant e-mail integration.
- Set/cancel call-forwarding and DND for other phones.
- Delay announcements capabilities to ensure callers are informed and provided other options during busy call times.
- Configurable night station or announcement when operators are not logged in.
- Flexible DID number assigned to each operator that follows them regardless of what PC/phone they use.
- Optional statistics package to track queue and operator statistics.
- Optional integration with Open*WorX* Location Status Information to see an employee's availability, return time, and alternate phone number.
- Optional integration with Open*WorX* Dialer providing Dterm users' access to the same employee and external database.
- Optional integration with Open*WorX* Message Reader providing Dterm users access OpenWorX text messages.
- Optional integration with Open*WorX* Desktop providing PC users all the benefits of the standard Dterm -based applications from a graphical user interface.
- FASTER and MORE EFFICIENT OPERATORS!

BAS: Call Control Functionality

Call control for the BAS is a queue-oriented system. Just like the NEC business attendant console, the BAS provides the standard familiar call queues and information. However additional information about incoming calls is added to better empower the Attendant.

Brief Definition of Queues:

- Internal – extension to attendant calls within the PBX.
- External – calls coming into PBX from an undefined trunk type.
- CCSA, FX, TIE, WATS – calls coming into the PBX from trunks in CCSA, FX, TIE or WATS Trunk Groups.
- Priority – calls that must be answered before all other calls.
- Paged – people who have been paged to a meet-me paging number, but the parked person is no longer waiting to be joined.
- Vacant – vacant-level intercept when a call goes to an invalid number.
- Park T.O. – calls that were parked and have timed out will recall here.
- NANS T.O. – internal calls blind transferred but not answered will recall here.
- Station Transfer – transferred to the attendant queue by another station or voice mail.
- Attendant Transfer – calls transferred from other attendants will arrive on this queue.
- Me – This queue is for the attendant's personal monitored number or calls made directly to their extension.

Brief Description of Call Control Function:

- Answer – answers the oldest call in queue or in the priority queue.
- Park – allows the operator to park a caller on an orbit number. Once there, dialing that orbit from any telephone in the system retrieves the caller. This function is also known as "Meet-me Paging".
- Dial – allows the operator to make an outgoing call from his or her extension.
- Retrieve – enables the attendants to retrieve parked calls by selecting a caller to retrieve from a menu of callers that are parked.
- Hold – place a caller on hold on one of the loop keys or "un-hold" them as well. The Dterm will become idle.
- Transfer – initiates a supervised transfer once a number has been entered or a directory entry selected.
- Blind Transfer – initiates an un-supervised transfer. As with Transfer, a number must be selected beforehand.
- Join Park – allows the operator to join an existing caller with a parked caller that is selected from the parked call menu.
- Conference – a conference between the source caller, the destination caller, and the operator.
- Source/Destination – used when a supervised transfer is initiated. Before the transfer is completed, the operator may switch voice path back and forth between the existing source and destination connections.
- Camp – camps a caller onto a busy extension.
- Cancel – disconnects a call or cancels an existing supervised transfer.
- Release – completes a supervised transfer or conference.
- Override – allows the operator to perform an executive override of a busy call.
- The above functions are mapped to Function Keys (F1 – F12) at the top of the keyboard. The spacebar can be used to answer the first call in queue and the Enter key will activate dialing or transfer based on the current call state.
- Other Functions and Features Associated With This Screen:
- Loop keys allow the attendant to hold up to 10 different callers simultaneously.
- The boxes directly below the loop keys are the source and destination boxes. The Source is on the left and is fully configurable to show the caller name and caller ID if known. Additionally, up to three lines of information chosen from the directory can be displayed below the name and number. The destination box is configurable in the same way and shows available information

when the operator transfers the call to another party, or when a station transfers a call to the operator.

- The Number Entered box allows the operator to enter a known number for either a direct dial or a transfer.
- The number shown in the bottom hand portion of the screen represents the number of calls waiting in queue at the queue screen, or total queue depth.
- The time shown left of the queue depth describes how long the current caller has been connected. Also if the operator hovers the mouse pointer over a loop key with a caller on hold, the time that caller has been holding will display.
- Left of the timer is an information screen that displays the name of the person holding while hovering the mouse pointer over a held loop. Also, if a caller is recalled from a park or a blind transfer, their identification, and the party to which they were transferred or for which they were parked will be shown here.

BAS: Speed Dials Pad

Up to forty speed-dial fields are available for each individual attendant. Dragging an entry from the directory onto the speed dial button automatically programs these fields. Another great feature of these buttons is that they act as a Busy Lamp Field (BLF). When that user's phone is off hook, the speed dial button becomes red and when that phone is ringing, the speed dial button blinks red. By right clicking the button, the operator can enable or disable monitoring, as well as change the name, forward the person's Dterm, set DND, or initiate a screened, blind, or announced voice call. This is useful for an operator who also has a set of VIPs for whom they answer calls.

BAS: Directory

Both an internal directory and an external directory exist in the BAS system. The internal directory lists employees within the company, while the external directory stores numbers outside the company. External numbers are usually numbers that are frequently called, or information about people or companies that frequently call the main listed number. When answering a call from one of these external numbers, fields from this entry are displayed to the operator, providing more information than just the 10-digit caller ID. There are several different fields available for display and four of these fields can be user-defined. The "Search" field searches dynamically as the information is typed in the field selected. The directory is also capable of being partitioned by company. The drop down box at the top of the window displays entries from a particular company, or all companies in the system. This is particularly useful in shared tenant installations, where a single attendant group serves multiple organizations.

DID for Company Lookup

A DID number can be associated with each company selection in the database. With this function enabled, when a call comes in on a specific DID for that company, the operator directory will automatically default to that company and only show those employees that are in the company. A screen pop will also occur letting the operator know which DID was called.

BAS: Overhead Paging

Most overhead paging is done by the creation of a direct trunk access key or a trunk access code. The BAS software does not have the ability to utilize these methods. Because of this, if the user wishes to overhead page from BAS they must manually initiate it from their Dterm. However the user may create a special speed dial key instead. In the PBX, the user must create a trunk access dial string. Using LCR, a dial string can directly access the trunk and provide voice path to the paging system. For example, the user creates the number 19 in their LCR to access trunk 1 in route 1. This trunk is terminated to the overhead paging. Now when the attendant hits their speed dial to dial 19, they will be able to overhead page.

BAS: Call Recording and Playback

Apart from the directory, another powerful feature integrated into the BAS client is the ability to record conversations and playback greetings over the telephone. With the addition of a Dterm Play/Record Module, the client will be able to automatically record voice conversations and instruct the software to save the last specified number of conversations. Conversations are saved as .wav files, which are time; date stamped and rotated in a circular buffer. When it is determined that a call needs to be saved, the operator simply presses the “Save Recordings” button and the selected call is automatically saved. Conversely, the operator can save a set of greetings to be played back immediately every time the phone is answered via the BAS. Greetings playback can be determined based on “internal” or “external” calls, on groups of caller IDs, or even morning or afternoon. A different greeting can be recorded for any of the scenarios above allowing the attendant to present a pleasant and consistent greeting.

BAS: Offline Mode (Pseudo Night Mode)

Night Mode allows for calls to be handled differently during off business hours. When no BAS attendants are logged into the system, the OWX server is still monitoring calls to the attendants. Since there are no attendants to send the calls, the server will instead send the calls to a configurable number, such as the voice mail system, or play a night announcement.

In the case where the server itself goes down, it is recommended that the user set up a UCD degrade. Since the OAI cannot route the calls, it will instead send the calls to a UCD group, which includes the operators. Typically a separate set of virtual sub-lines are put into a UCD group and assigned on each attendant’s Dterm and the pilot number of this UCD group is associated with the BAS queues to be used in the event of failure.

BAS: Authorization and Account Code Dialing

BAS supports dialing with Account Codes or Authorization Codes. However, it uses a slightly different number format for the dial string. To denote an account code an exclamation point (!) is used and for authorization codes a dollar sign ($) is used. Here are some example strings.

Account Code: 2145551212! 1234567890
Authorization Code: 2145551212$ 1234567890

The exclamation point (!) or the dollar sign ($) delineate to BAS where the dial-able number ends and where the code begins. In the account code example, 2145551212 is the phone number and 1234567890 is the account code.

Open*WorX*: Business Receptionist (BR)

The Open*WorX*® Business Receptionist is a PC-based call-processing application designed primarily for departmental executive assistants. The Open*WorX*® Business Receptionist enables multiple users to answer calls, search the directory for a desired party, check the Dterm status, and perform call processing functions including making, answering, transferring, and parking calls. The Open*WorX*® Business Receptionist is a line-based system unlike the Business Attendant System (BAS) which is queue-based. Therefore, the Open*WorX*® Business Receptionist is also an effective tool for other Dterm users such as help desks, security desks, etc. The OpenWorX Business Receptionist, like all Open*WorX*® applications, uses the common Open*WorX*® database and is integrated with a multitude of Open*WorX*® applications.

BR: Speed Dials Pad

The BR speed dial pad is modeled off the BAS speed dial. See the previous section on BAS for details of operation.

Note: *In order for the BR operator to monitor the speed dials, they must be given TAPI rights to the lines they wish to monitor*

BR: Directory

Both an internal directory and an external directory exist in the Business Receptionist system. The internal directory lists employees within the company, while the external directory stores numbers outside the company. External numbers are usually numbers that are frequently called, or information about people or companies that frequently call the main listed number. When answering a call from one of these external numbers, fields from this entry are displayed to the operator, providing more information than just the 10-digit caller ID.

DID for Company Lookup

The directory is also capable of being partitioned by company. The drop down box at the top of the window displays entries from a particular company, or all companies in the system. This is particularly useful in shared tenant installations, where a single attendant group serves multiple organizations. Additionally, a DID number can be associated with each company selection in the database. With this function enabled, when a call comes in on a specific DID for that company, the operator directory will automatically default to that company and only show those employees that are in the company.

BR: Overhead Paging

Most overhead paging is done by the creation of a direct trunk access key or a trunk access code. The Business Receptionist software does not have the ability to utilize these methods. Because of this, if the user wishes to overhead page from Business Receptionist they must manually initiate it from their Dterm. However the user may create a special speed dial key instead.

In the PBX, the user must create a trunk access dial string. Using LCR, a dial string can directly access the trunk and provide voice path to the paging system. For example, the user creates the number 19 in their LCR to access trunk 1 in route 1. This trunk is terminated to the overhead paging. Now when the attendant hits their speed dial, to dial 19, they will be able to overhead page

BAS: Call Recording and Playback

Apart from the directory, another powerful feature integrated into the Business Receptionist client is the ability to record conversations and playback greetings over the telephone. With the addition of a Dterm Play/Record Module, the client will be able to automatically record voice conversations and instruct the software to save the last specified number of conversations. Conversations are saved as .wav files, which are time & date stamped and rotated in a circular buffer. When it is determined that a call needs to be saved, the operator simply presses the "Save Recordings" button and the selected call is automatically saved. Conversely, the operator can save a set of greetings to be played back immediately every time the phone is answered via the Business Receptionist. Greetings playback can be determined based on "internal" or "external" calls, on groups of caller IDs, or even morning or afternoon. A different greeting can be recorded for any of the scenarios above allowing the attendant to present a pleasant and consistent greeting over the telephone.

BR: Authorization and Account Code Dialing

Business Receptionist supports dialing with Account Codes or Authorization Codes. See the previous section on BAS for details of operation.

BR: Attendant PC Minimum Hardware Requirements

Business Receptionist hardware requirements are the same as the BAS requirements, see previous section for details.

Open*WorX*: Communications Portal

Business communications in the 21st century is no longer defined strictly as phone calls and documents. It has become pervasive in the business day, taking many forms – phone calls to the desk, phone calls to cell phones, emails, voice mail, faxes and instant messaging.
The Open*WorX*® Communications Portal provides the added value of including communication tools within the portal. This enables information technology organizations to provide their users with a single interface for multiple communication applications. The consistent interface across the organization can streamline administration, management and user training while providing the greatest level of functionality to users.
Many daily business activities rely on communications, but at the same time, business people can easily be overwhelmed by the volume of contacts – and the variety of communication forms enables anyone to interrupt at any time. Today's business professional must be empowered to manage their communications. Open*WorX* Communications Portal integrates all of these channels into a unified, web accessible portal that puts the individual back in control of their own communications. All contacts can be simply and easily managed from a single application – reducing the complexity of dealing with multiple communication types and increasing personal productivity.
Today's business person is mobile, yet they are dependent on communications to maintain the flow of their business. Most business communication attempts fail on the first attempt because the receiving individual is away from their desk or is otherwise engaged. The Open*WorX* Communications Portal will understand the individual's status. Using flexible rules and capitalizing on the integration of voice and data, it will selectively route contacts and messages based on the identity of the sender as well as the status of the individual. This allows the

business user to define who they want to interact with in real-time – to complete those critical projects – and who they would prefer to "get back to" at a later time.

Communications Portal: Functionality

Portal

The Portal functionality was designed to grant the individual user access to enterprise web-based backend systems such as databases, HR applications, and existing business portals. The Portal supports custom or company created portlets, so tools can be provided to users to help them better perform their daily business tasks. Users can also add Internet sites and applications, personalize their settings, edit screen layouts, and change color and title schemes.

My Contacts

Having the ability to find the people you need to contact is imperative. The "My Call List" combines call logs, a custom contact or buddy list – this list can be comprised of individuals or groups – along with personal and corporate directories in one application. This makes it fast and easy to find that important contact. The personal directory can be synchronized with Outlook contacts, and the corporate directory links with the OpenWorX directory via LDAP.

Once you locate a contact, "My Call List" makes it quick and easy for you to connect. With a single click you can make a phone call, launch an email, start a voice, video or collaboration session, or initiate an instant message. You simply choose the contact vehicle and launch it directly from a custom made contact list, or from any directory.

Communications Portal logs incoming and outgoing calls, whether dialed using the Communications Portal, or by hand. You can sort all call logs by Name Number or Time, and easily return the call by clicking on the number.

My Controls

The "always available" tool bar interface of "My Controls" gives the user the flexibility to change status and contact numbers on the fly. The "always available" tool bar interface of "My Controls" gives the user the flexibility to change status and contact numbers on the fly.

The user can make a call using the number entry box in the "My Control" toolbar. They don't have to enter 9 or 1, the OpenWorX dialing rules take care of it automatically. These intuitive controls give the user the functionality they need to stay in touch, and be more productive in a fast paced business environment.

Contact Rules and Methods

Communications Portal employs user based contact rules and methods. Users can create methods for each status and by the device they want to be contacted by, depending on who is calling them. Contact rules and methods can be customized by the user to reflect how they wish to be contacted. Up to 6 different methods can be established for any status and each rule can be assigned to any caller or groups of callers. Outside calls are automatically routed to the next device, and inside callers can forwarded the call to the next available device.

My Outlook

The portal mail client is integrated with MS Outlook. This gives the user the ability to manage messages and other personal information without leaving the portal. Calendar link controls are built-in to automate the user's status by scheduled events. "My Outlook" synchronizes with additional Outlook functionality, making it easy to initiate tasks and manage your mail.

Communications Portal: Benefits

- Improve efficiency by combining all telephone and messaging activities within a single application

- OpenWorX® Communications Portal allows staff members to manage all of their communications needs within a single window. This allows individuals to place calls, return calls, send and receive email, manage their contact information and check messages without having to switch applications. Fewer applications streamline work processes and minimize confusion.

- Enhance customer satisfaction with call management
  - Staff members can identify calls from priority customers and have them routed based on the source, giving the individual the flexibility to handle these calls quickly, even when away from the office. Key individuals remain available to callers regardless of their location, so mobility isn't a hindrance to productivity.
- Improve organizational productivity
  - The ability of the OpenWorX Communications Portal to track location status can help the entire organization stay better connected. The ability to locate key personnel who are often away from their desks for a variety of reasons for a period of time can keep processes within the organization moving to enhance everyone's productivity.
- Improve communications for remote workers
  - OpenWorX Communications Portal is browser-based and requires network access for full operation; it is an ideal application to help keep remote staff members in touch with the rest of the office. Remote users can have all the same capabilities as local users, including managing contacts, setting and viewing location status information and use of both corporate and personal directories.

OpenWorX: Dialer

NEC's OpenWorX Dialer allows any user in the UNIVERGE NEAX 2000 IPS to use his or her Dterm to electronically search for the extension of another person in the UNIVERGE NEAX 2000 IPS, even while the user is on the phone. Once the listing is located, a simple button press is all it takes to make the call or initiate a transfer. Instead of thumbing through a paper directory, the user now accesses the online up to date directory by pressing a function key on their Dterm. Centralizing the database also ensures that all extensions are easily added and/or updated; no matter how often updates are made. Also, the user can query using division and department such as would occur in a multi-site corporation.

Dialer: Database

The database used for Dialer is the same as the Business Attendant System database, which allows Dialer users to have the same information available to the attendants.

Dialer: Benefits

- Saves the attendant time by curtailing the need for employees to dial '0' to get a phone number or extension of someone in the company directory.
- Provides a centralized company phone directory.
- Eliminates outdated printed copies of the company directory.

Open*WorX*: Location Status Information (LSI)

LSI is a useful service that allows other system users to obtain information regarding your current location and status when you are away from your office. This can be done through any Dterm phone that has the associated function key programmed. The user can:

- Set, Modify, or Delete their location information.
- View the location information of your colleagues.
- Query an extension to determine whether location information is set, and call that extension, if the person is available.
- Check the status of other users simply by dialing their extension, if configured to do so.

LSI: Database

LSI uses the Open*WorX* database. This common database allows for other applications such as BAS, INASET or Desktop to access LSI information.

LSI: Benefits

- Never have to guess about someone's whereabouts.
- Know when someone is expected to return and how to contact them while they are away.
- Securely set your own location through any Dterm phone in the office.
- BAS integration allows attendants to check status via their directory.
- Desktop integration allows all employees to view the status of other employees via their web browser.
- INASET integration allows INASET users to see employee statuses in their view of the corporate directory.

Open*WorX*: Message Reader (MR)

Message Reader is a convenient application that allows users to retrieve their messages taken by the attendants from any Dterm phone set enabled with the proper OAI function key. The messages can be sent from BAS, INASET or Desktop.

MR: Functionality

Message Reader is quite easy to use. Simply go to any Dterm phone set that has the Message Reader function key configured, then press the key and enter the extension for which you wish to read messages. In the event more than one user is assigned to a phone, the user may choose from a list of employees to designate for whom to read messages, and then enter the proper password, if required. If no extension number is entered, Message Reader will default to the extension you are using for this connection. Once having entered the extension and the password, the user may easily peruse all their messages and after a message is read, it may be deleted or saved. If the person leaving the message left a return phone number, Message Reader can even dial the call back number automatically!

MR: Database

MR uses the Open*WorX* database. This common database allows for other applications such as BAS, INASET or Desktop to access LSI information.

MR: Benefits

- Read your messages even when away from your desk, from any Dterm phone set.
- Save valuable operator time because they no longer have to read the message back to the employee.
- Automatically dial call back numbers, preventing paper clutter or misdialed numbers.

Open*WorX*: Short Text Messaging

Short Text Messaging (STM) is a service running on the Open*WorX* Server designed to facilitate the delivery of text messages to NEAX Wireless Dterm PSIII handsets on the *UNIVERGE* NEAX 2000 IPS. This is accomplished with either, Simple Mail Transfer Protocol (SMTP) on a Local Area Network (LAN), or Telelocator Alphanumeric Protocol (TAP) Version 1.8. This application turns your PSIII handsets into one-way internal text pagers in addition to their normal functionality. The interface allows for an audible alert when messages arrive. Users should be aware that the volume for the text message arrival notice is the same as the ringer volume. In addition, if the PSIII is set to vibrate, it will also vibrate when a text message arrives. Messages can be stored for later reading as well as deleted. Messages cannot be received while the user is on the phone.

STM: Functionality in an SMTP Environment

The STM server can be installed on your enterprise network as an SMTP mail gateway. In this configuration, users simply send messages up to 128 characters (always the limit of text messages, no matter what SMTP set up used) to the address PagerID@domain.com. 2000@wireless.user.com is a good example. The server will then parse the Pager ID out of the heading and send the message to the wireless handset with that extension. NEC has verified this functionality on Microsoft Exchange® and Lotus Notes®.

STM: Functionality in a TAP Environment

STM can also be configured to use an RS-232c TAP interface for sending or receiving alphanumeric messages to the user's wireless handsets. Thus a TAP enabled pager program can be used to send a message.

STM: Benefits

- Allow users to send messages directly to users when not available for call transfer.
- Short messages can be left as text pages instead of voice mails.
- Internal Text messages can be sent directly to wireless handsets via email.
- Facilitates quick call back. "Call me at 555-1212, Bob." Can immediately be returned via the same device.
- Automated reminders can be set up via your email to be sent directly to your PSII.

Open*WorX*: Incoming Call Assistant (ICA)

Incoming Call Assistant (ICA) is a service that gives Dterm users a powerful tool to manage incoming and outgoing calls. These functions can be accessed with the simple touch of an associated function key.

ICA: Functionality

The user can:

- View the caller's name and number on the LCD display for any line or sub-line on the user's Dterm, even while currently on another phone call. The name lookup comes from the Personal, External, or Corporate Directory in OpenWorX. The above order is also the search order priority.
- Check for past or current calls to the user's Dterm on any Dterm in the system with an ICA enabled function key.
- Look up past callers and return their calls with the information stored in ICA's database. This is known as the inbound call log. It will store up to 99 entries.
- Check list of calls made by the user. This is called the Outbound Call Log. Up to 99 entries will be stored.
- Supports use of * and # in the user extension with the limitation that the user must be at their physical device to log in if their extension contains either a * or a #.
- Determine if a call was forwarded or transferred.

ICA: Functionality

ICA uses the same database as the entire OpenWorX suite of applications. The following fields from the OpenWorX Database are accessible.

- Caller name (inbound and outbound call log)
- Caller number (inbound and outbound call log)
- Date and time of each call (inbound and outbound call log)
- Whether or not the call was transferred and by whom (inbound call log)
- Whether or not the call was forwarded and by whom (inbound call log)
- The direct number dialed by the caller (inbound call log)

ICA: Benefits

- Able to identify callers on prime lines and sub-lines before they are answered, even while the user is still on another call.
- Identify past callers quickly without having to listen to voice mail.
- Identify past callers when you are not at your desk.
- Return calls quickly with dial back feature.

ICA: Comparison of ICA and IPS Functionality

Feature	OpenWorX ICA	2000 Caller ID
Call Log Capacity	99 Inbound max 99 Outbound max (System Level Setting)	24 Inbound, or 19 Inbound + 4 Outbound
Dialed call types saved in Call Log	Manual, TAPI, OAI, LNR	Manual
Call Log Information displayed	Number, Name, Date, Time, Transferring/Forwarding number, Line rung	Number only
Ringing Line Information Displayed	Automatic, Name & Number	Prime Line- Automatic Sub line – Manual, key press shows Name & Number
Caller ID “Name”	Searches Personal, Corporate, and External directories for matching name to Caller ID number	Displays Caller ID name and number form Telco. CID name can be overwritten by system defined name for 1500 numbers

ICA: ICA vs. Camp On

It should be noted that ICA and Camp On have similar end goals. With camp on, the user receives notification of a second call that is “camped” to their extension. They have the choice of answering this call and putting the original caller on hold. However, Camp On is limits the user to 2 total callers.

ICA can achieve the same thing through multiple sub lines. The user rolls their forwarding of busy calls to the next sub line. ICA then gives notification of the information of the incoming caller. From there the user can decide whether or not to answer the call or let it continue and roll to voice mail.

Open*WorX*: Group Call Forward Control (GCFC)

Group Call Forward Control is an application that allows forwarding patterns of individuals or user defined groups to be forwarded based on a schedule. Complex schedules are supported, that allow for recurring forwarding on a daily, weekly, monthly or even yearly basis. GCFC supports call forward busy, call forward no answer and call forward all calls. It is possible to forward entire divisions within a company, such as after 5:00 PM all calls are forwarded directly to voice mail.

GCFC: Functionality

The user can navigate completely through all the tasks required of the GCFC administrator with a simple point and click of the mouse. The administrator can also see a summary of what is already configured within the application. The administrator can modify, define and delete user accounts here. There are several levels of users.

- Administrator – can access and maintain all user accounts, groups and call forwarding schedules within GCFC.
- Premium User – can create and maintain sets of groups. Including adding, modifying and deleting group members, as well as controlling their call forwarding schedules.
- Regular User – can view group and call forwarding jobs within GCFC, as well as maintain a group if given permission by the group owner

The users of the groups can also be given or denied permission to group level. There is no limit to the number of users to each individual group, nor is there a limit to the number of people who can modify the group members or call forwarding patterns. Jobs can be configured to occur only once or recur at specific time intervals. These time intervals can be as small as forwarding all calls to voice mail at 5:00 PM every weekday, to as broad as forwarding all calls to an automatic announcement for holidays that occur once a year

GCFC: Functionality

- Allows calls for individuals or groups of people based on schedules.
- No more forgetting to un-forward your phone after returning from vacation or a business trip.
- Insure privacy for users, such as in a hospital or dormitory.

Open*WorX*: Personal Call Assistant (PCA)

In NEC's continuing effort to bring better efficiency to the standard telephony user's desktop, Personal Call Assistant was created. PCA includes functionality like Desktop and INASET in a much more convenient form. PCA replaces Clipboard Dialer (the ability to dial from any application that supports the Windows Clipboard). PCA brings screen pops to the user desktop, via text to speech agent, small pop box, or specially written customized pops. The text to speech agent and small pop box are standard features. The customized pops must be specially written. It is recommended that PCA's server reside as an intranet application with clients treating it as a Trusted Site in their internet options.

The icon is the same as Clipboard Dialer, a small yellow phone in the system tray. This icon will turn red when the user has missed calls. Hovering over the red icon with your mouse will list the number of calls that have been missed.

The user can then access their inbound call log to see a list of the calls that have come in to their phone The name of the caller will be there if that person is in the OWX database. The inbound number, whether extension or caller id, is recorded as well as the time and date of the call. A day of the week will be shown for the last 7 days of calls. Calls going back more than 7 days will show a date such as 1/10. Clicking on the call number will return the call with the correct dialing rules. The user can also check their outbound call log in the same way.

If the user simply wishes to find someone internal to the company they can use the corporate directory. Clicking on the telephone icon next to the person will dial that person. Dialing rules are applied automatically. Other numbers can be stored and displayed this way, as well as email and pager. These fields just like Desktop can be masked from certain users. Searching can be done by entering a letter combination and hitting GO. You can search by Full Name, Last Name, First Name, or Extension. In addition to the corporate directory, each user gets a Personal Directory. This directory is stored on the server, but is only accessible by that user and operates the same way as the Corporate Directory. The Personal Directory will also support importing from a CSV (comma separated values) text file. PCA also includes the ability to synchronize the user's Personal Directory with their Outlook Contacts. The mappings of the fields are controlled by an XML file (mappings.xml in the PCA directory) which is easily modified to suit the individual's needs.

PCA: Limiting User Views

Occasionally it is necessary to make certain fields un-available except by certain users. For example, the company may not wish for non-executives to have access to employee home phones. Because of this, the Open*WorX* administrator can put the login Ids of Desktop users into groups that do not allow the viewing of specific fields.

PCA: Open*WorX* Desktop

Desktop is now a part of Personal Call Assistant as a standard feature. Desktop is designed for general day-to-day *UNIVERGE* NEAX 2000 IPS users who have a PC with a web browser. No application installation needs to take place on the PC. Access is directly via the web.

Desktop Directory

The user can look up individuals within the company directory, and access their speed dials. From each, they can dial, and from within the corporate directory, a user may send a Short Text Message (if STM is installed), email, or OpenWorX Message.

Personal Directory

In addition to the standard Open*WorX* corporate directory, Desktop gives users an additional Personal Directory. The Personal Directory is different for each individual user, but the Corporate Directory is common among all users. For example, the user could have an entry for their Mother, and it would only be for use with their individual phone.

Desktop Call History

The lists of calls that have been received by the phone are displayed in the "Inbound" Section of this screen. The number, Name, and Date are included. The Name is only included if there is a match for the number in the database. The user can also click on the telephone handset icon next to the number to return the call from their Dterm set.

In addition to searching the corporate directory, ICA will now search the Personal Directory of that station as well. For example, if 214-555-1111 calls one individual who has "Mom" defined in their Personal Directory for that number, the ICA pop and record in Desktop will say "Mom". For another person who does not have "Mom" defined in their Personal Directory, no name will be attached, or can also have a different number defined for "Mom. ICA also indicates the phone type found in the Personal contact, using appending "-H" for home, "-M" for mobile or wireless, and "-W" for any business line.

The "Outbound" section is similar to the "Inbound" section, except it records calls made (by Manual Dialing, Open*WorX* Dterm Applications, Desktop, or TAPI) at the telephone, not calls received.

Desktop Location Status
With Desktop, the user can set their status, including return date, return time, and alternate number with the click of a mouse. In addition, Desktop allows for the use of Custom statuses, instead of only the 10 pre-defined messages available via Dterm operation.

Desktop Message Reader
Previous to Desktop, the only way for an individual to check messages sent by Message Reader was by using the key on the Dterm, or by calling the attendant and having the message read back to them. Now, the user can use a Web Browser with Desktop to check their messages. Also, as mentioned in the Directory section, the user may send these messages to other users of Message Reader. No longer is it just the attendants who can send messages. The message can also include a number and the Desktop user simply needs to click on the telephone handset icon next to the number to return the call from their Dterm set.

PCA: Open*WorX* INASET

The INASET is now a part of Personal Call Assistant as a standard feature. NEC's INASET is a powerful IP terminal and is enhanced even further by the addition of the OpenWorX application suite. This suite enables INASET Application Services mode, adding a great deal of features and functionality.

Home Application
This is the base application that is loaded directly to the INASET's memory. It allows for the basic functionality expected from a standard IP Dterm. The main difference a user will notice with this screen is the different application control buttons at the bottom of the INASET screen. Without Open*WorX*, these buttons are static and controlled by the INASET's local memory. With the addition of the OpenWorX Network Services mode, the server configures these buttons. The modification of a simple XML file on the server is all that is required to move, add, or delete additional application buttons. These additional application buttons can be used for many things, such as custom developed applications and direct links to web sites.

Corporate Directory
The Corporate Directory pulls the information directly from the Open*WorX* Employee database. The user can search for specific people and dial using this directory. Once found in the directory the user may also send a text message (if Message Reader is installed), send a Short Text Message to their wireless PSIII (if Short Text Messaging is installed), or simply view the entry to see other information or alternate dial-able numbers.

Location Status
This interface allows the user to set their status in the OpenWorX database. It is then accessible by the operator. Also when another Dterm user calls the person, they will receive their status on their Dterm display, along with Return Time, and Alternate Number if provided.

Message Reader
Message Reader allows the user to read messages sent to them by the operator or other users.

Open*WorX*: Name Display

Name Display is an application that synchronizes the *UNIVERGE* NEAX 2000 IPS name database with the employee portion of the OpenWorX Database. This sync is real time.

In the case of two people sharing a single extension, the application will search through, Department, Division, and Company in that order. The first one that matches for both employees will be the one displayed.

Open*WorX*: Multiple CCIS Node Configuration

OpenWorX is capable of hosting applications for multiple *UNIVERGE* NEAX 2000 IPS systems on a single server. All instances of this require a viable OAI connection between the OpenWorX server and the *UNIVERGE* NEAX 2000 IPS. See the bandwidth requirements list previously in this chapter to make sure your connection is sufficient.
The applications once equipped with the remote TCP/IP connection and multi-node software operate as if they are in a single *UNIVERGE* NEAX 2000 IPS, except where noted below.

BAS: CCIS Network Information

The Business Attendant System can run in a CCIS network environment with the following limitations.

- Calls transferred from the BAS Client across a CCIS Link will not recall to the BAS operator.
- Using multi-node capabilities, the operator can now view station data via the directory or speed dials to extensions in other IPS systems. The other IPS systems must have OAI and a TCP/IP connection back to the Open*WorX* server. Additional Open*WorX* software is also required per IPS to be added on the server.

Business Receptionist: CCIS Network Information

The Business Receptionist can run in a CCIS network environment with the following limitations.

- Calls transferred from the Business Receptionist across a CCIS Link will not recall to the operator.
- The Dterm across a CCIS Link cannot be monitored by the Business Receptionist system and the monitored speed dials and status checks in the directory will not function for those extensions.
- The system will force supervised transfers to CCIS extensions to ensure callers do not get transferred to a busy remote extension.
- The message-waiting lamp cannot be controlled on a remote Dterm.

Note: *Business Receptionist does not have the same multi-node capability that BAS does*

OAI Application Software Development

For users wishing to develop their own applications, NEC offers a complete library of development tools. For further information, contact the OAI Product Manager.

System Specifications

Required Equipment for OAI

Equipment	Description	Qty	Remarks
PN-CP24-C (MP)	MP Card	1	One card per system Mount on PIM0
PZ-M606-A (ETHER)	Ethernet Control Card	1	Mount on MP card One card per MP
• 10BASE-T Cable • 100BASE-TX Cable	10BASE-T/100BASE-TX Cable between Ethernet Card and LAN	As required	Cable length: Maximum 100 m (328 ft.)

Chapter 12 Automatic Call Distribution (ACD) with MIS

Automatic Call Distribution (ACD)

An Automatic Call Distribution (ACD) feature permits incoming calls to terminate to a prearranged group of stations. Calls are distributed in the order of arrival to idle terminals within the group, based on which terminal has been idle the longest period of time. Stations may log on/log off from the ACD group. Supervisor stations may monitor conversations of agents.

In addition to basic ACD, NEC has a Microsoft® Windows 2000 Professional, Windows 2000 Advanced Server or Windows 2003 Server ACD/MIS system (CallCenter*WorX* ACD/MIS) that enables a user to efficiently manage a call center. NEC offers an Infolink Application (Que*WorX*) that provides IVR, Internet and Desktop integration for *UNIVERGE* NEAX 2000 IPS Call Centers. Finally, Q-Master which offers a modular call center environment allowing for more flexible options by choosing just the application needed.

The MIS system analyses statistical data received from the ACD, displays real-time activity screens, and generates and prints graphs and reports of the activity.

Basic ACD

The *UNIVERGE* NEAX 2000 IPS has basic ACD that comes standard with any basic system package.

Service Conditions

1. A maximum of 16 ACD groups can be assigned per system. Each ACD group is assigned a pilot number. Calls directed to the pilot number are directed to that ACD group.
2. The maximum number of stations in an ACD group is 60. The maximum number of ACD groups in the system is 16. If ACD-MIS is used, the maximum number of ACD stations is 60.
3. Assignment of ACD groups is performed from the Maintenance Administration Terminal (MAT) or Customer Administration Terminal (CAT).
4. ACD groups consist of a pilot station and one or more member stations. Hunting is initiated in a circular fashion, and then based on which member has been idle the longest period of time.
5. If all stations within the ACD group are busy, incoming calls may be serviced in the following ways:
   - remain in queue until an agent becomes available (Ring back Tone provided)
   - immediately overflow to another group, to a station, or to the Attendant
   - remain in queue until an agent becomes available (Delay Announcement or Music on Hold provided)
   - remain in queue for a preset time (Ring back Tone, Delay Announcement, or Music on Hold provided), and then overflow to another group, to a station, or to the Attendant
6. When the pilot station has set Call Forwarding – All Calls, incoming calls to the ACD group will be transferred to the destination of that Call Forwarding – All Calls setting

Service Conditions (Cont)

7. An ACD group number can be used as the destination station of Direct Inward Termination (DIT), or as a designated Night Service station.
8. An ACD group number can be assigned as the destination station of Off-Hook Alarms, Priority Calls, and Attendant Night Transfer.
9. ACD group pilot numbers should not be placed in Station Hunting groups. The Station Hunting feature would take priority over the ACD function.
10. Two types of traffic measurements can be provided for ACD:
    a. ACD group Peg Count
        - Count of incoming calls
        - Count of answered calls
        - Count of abandoned calls
        - Count of waiting calls
        - Count of all busy calls

    b. ACD station Peg count
        - Count of answered calls
11. Upon initial installation, or after a system initialization (reset), each agent must lift and restore handset (of their station) to begin receiving calls for the ACD group.

Basic ACD Features

Busy In/Busy Out – ACD

This feature allows an agent in an ACD group to log their station onto or off of the group. This allows the system to control whether a call directed to the pilot number of the ACD group goes to that station or not. This prevents incoming calls from being directed to stations at which no agent is available.

Call Waiting Indication – ACD

This feature provides a visual indication when an incoming call to an ACD group is placed in queue, due to an “all agents busy” condition. On external relay controlled indicator or an LED on a Multiline Terminal can be used to provide Call Waiting Indication.

Delay Announcement – ACD

This feature allows the system to provide a recorded announcement to an incoming caller placed in queue to an ACD group. A single announcement, or two separate announcements, can be provided.

Hunt Past No Answer – ACD

This feature allows calls targeted at an ACD group to hunt past an agents station, after a no answer condition, if the agent forgets to log off of the group and the agent is unable (or not available) to answer the call.

Immediate Overflow – ACD

This feature allows a call directed to an ACD group to immediately overflow to another ACD group, upon encountering an "all agents busy" condition.

Priority Queuing – ACD

This feature allows the system to prioritize incoming calls by trunk route and on a per station basis, when the call enters an ACD queue. When a call is a considered as priority it is placed at the beginning of the queue.

Queue Size Control–ACD

On incoming DID/Tie line calls the system can be assigned a threshold that limits the number of calls in queue. When the queue size threshold is exceeded, incoming callers are connected to busy tone.

Silent Monitor – ACD

This feature provides the ACD group supervisor with the ability to monitor a call to an ACD agent. The silent monitor function gives no indication (as an option) to either the agent or the calling party.

Note: *The use of monitoring, recording, or listening devices to eavesdrop, monitor, retrieve, or record telephone conversations or other sound activities, whether or not contemporaneous with its transmission, may be illegal in certain circumstances under federal or state laws. Legal advice should be sought prior to implementing any practice that monitors or records any telephone conversation. Some federal and state laws require some form of notification to all parties to the telephone conversation, such as using a beep tone or other notification methods or require the consent of all parties to the telephone conversation, prior to monitoring or recording a telephone conversation. Some of these laws incorporate strict penalties.*

CallCenter*WorX* ACD 3.0 for Business

CallCenter*WorX* ACD 3.0 for Business is a suite of hardware and software solutions that are focused on customer care. Included are the Automatic Call Distribution (ACD) system and the Management Information System (MIS), elements that streamline call flow and workflow. Its routing capabilities allow businesses to connect callers with the most appropriately skilled agents in the shortest time possible, minimizing a caller's wait for assistance and improving customer satisfaction. Its real-time and historical reporting abilities provide call center managers and supervisors a tool for maximizing agent productivity, assuring quality performance in handling the incoming and outgoing call volume in the call center. CallCenter*WorX*-Business helps businesses make a good first impression on new customers, better manage existing customer relationships, and inspire customer loyalty, leading to increased revenue potential.

CallCenter*WorX* ACD 3.0 for Business is a Microsoft Windows® based software system that enables you to efficiently manage your call center. It does so by providing a complete and flexible feature set, allowing you to customize the ACD based on your particular business needs. In addition, it provides a powerful Graphical User Interface (GUI) to a *UNIVERGE* NEAX 2000 IPS ACD System that allows you to program additions, deletions, and changes via your computer. You can also query and print ACD data from CallCenterW*or*X.

The CallCenterWorX ACD 3.0 for Business suite includes the Automatic Call Distribution (ACD) system and the Management Information System (MIS 3.0). The ACD system supplements the call processing capabilities of the PBX, with typical call routing designed to connect each caller with the most appropriate agent to handle their call in the shortest time possible.

When the CallCenterWorX ACD system accepts a call, it can be configured to play a message to the caller and route the call in a pre-specified order to a queue of waiting calls for a particular split (group of agents). The queued call is then answered in sequence by the next available agent in that split. The Call Control Vectors of the CallCenterWorX ACD system allow great flexibility in routing calls, permitting alternate solutions to fit almost any situation.

As each call progresses through the system, the CallCenterWorX ACD system provides data to the CallCenterWorX MIS 3.0, which translates the call-handling data into meaningful statistics. The MIS uses these statistics to offer call center managers and supervisors access to real-time and historical data concerning incoming and outgoing call volume.

The CallCenterWorX MIS Status Screens (Real-time Screens) provide color-coded, up-to-the-second views of agent, queue, and system activity. Statistics on the call volume and agent effectiveness are computed in real-time and are displayed for the current hour and day on supervisors' screens. Optional wall display boards can show current statistics and messages to call center personnel.

The call statistics are also collected in the MIS database and used to generate reports in text or graphical format. These reports and graphs can be printed for a view of activities ranging from a graph updated in the last second to a retrospective report ranging back up to 1.5GB worth of data.

CallCenterWorX ACD Features

1. Runs on the Windows 2000 Professional operating system using Client/Server architecture, making information and processing directly available from the desktop interface.

    - The MAT Client as a remote PC runs on Windows 2000 Professional or Windows XP.

2. A capacity of 10 to 120 positions, depending upon licensing, application, and PBX platform.

3. Can support up to nine tenants.

4. Uses a GUI interface to make commands and controls easily accessible to the user.

5. Routes incoming calls using:

    - User-defined sets of call handling instructions named Call Control Vectors (CCVs), week schedules, and holiday schedules.

    - A system of priorities and call overflows designed to deliver calls rapidly and efficiently to the appropriate split or agent.

6. Has dock able toolbars, which can be dragged to other locations on the working screen. The user can also build a personal toolbar of most frequently used icons.

7. Provides toolbar icons and tool tips for all MAT commands, for opening report views, and for online Help and context-sensitive Help.

8. Provides data entry screens with a choice of background color pattern for each screen.

    - Data entry screens can accept new data or revise previously entered data with immediate confirmation of the action. These screens list operational data in drop-down menus.

    - Each data entry screen can open to a report view showing the currently programmed data. The font style and size and the column headings in the report can be reset to suit your preference.

    - The report view can open to a print preview showing the report as it would be printed, and can send the report to a printer. A report can be saved to a data file, saved to a Microsoft Excel file, or exported to a Microsoft Access database.

9. An optional workbook area showing a tab for each open screen with the MAT command icon for that screen. A report icon on top of the command icon for those screens open to the report view.

    - Multiple screens can be open simultaneously. A screen is made the active screen by selecting the appropriate Workbook tab.

10. Has an optional status bar, which reflects the state of the active screen.

11. Supports 6 languages (English, Japanese, French, German, Italian and Spanish) for display to the agent's Dterm position.

12. Uses a Trace Client to view ACD call processing on screen. CCV programming diagnostics let you monitor the call flow in real-time to design or re-engineer your installation.

13. ACD client MAT provides the ability to program multiple CallCenter*WorX* ACD`s from one GUI in a network

14. List-up includes the ability to view multiple reports and to export these reports in MS Access or MS Excel formats

15. Configuration of trunk group data and holiday calendar performed by single point and click interface

16. System Data configuration allows the ACD to be customized for specific features including Info link and MIS message generation, debugging tools, and phone-set operation

17. Trace Client view provides the ability to see ACD call processing on screen. Call Control Vector (CCV) programming diagnostics allow the call flow to be monitored in real-time, aiding in the design or re-engineering of call center installations

18. A Windows® event viewer provides access to critical ACD event messages such as ACD state changes and helps technicians identify other critical events such as an inability to connect to the PBX or indicating that the system security key is not attached

CallCenter*WorX* MIS

The CallCenter*WorX* Management Information System (MIS 3.0) offers call center managers' access to the most important real-time and historical data available for workgroups. With this tool, you are able to maximize agent productivity and assure quality performance in handling the incoming and outgoing call volume of the center.

CallCenter*WorX* MIS has a menu-driven user interface. All of the menus show a list of choices available under a particular topic. Each choice is preceded by a number, which is entered at the Enter Selection prompt, followed by pressing the Enter key. CallCenter*WorX* MIS responds by displaying the requested menu screen, by using one of its prompting methods to ask for more information, or by displaying the appropriate data screen.

CallCenter*WorX* ACD 3.0 for Business Turnkey Software Only is available packaged for 10, 20, 30, 40, 60, 80, 100, and 120 Agents. Each combines both the ACD and MIS software on a CD ROM and comes with a required security key for the main CallCenter*WorX* PC. The security key attached to the main CallCenter*WorX* PC provides the appropriate agent capacity as well as the main CallCenter*WorX* supervisor position. Individual security keys can also be purchased for applications requiring additional supervisor PCs.

CallCenter*WorX* ACD 3.0 for Business Turnkey Packages

Business Turnkey Packages
CallCenterWorX-B ACD (3.00) (10 Agent Pkg)
CallCenterWorX-B ACD (3.00) (20 Agent Pkg)
CallCenterWorX-B ACD (3.00) (30 Agent Pkg)
CallCenterWorX-B ACD (3.00) (40 Agent Pkg)
CallCenterWorX-B ACD (3.00) (60 Agent Pkg)
CallCenterWorX-B ACD (3.00) (80 Agent Pkg)
CallCenterWorX-B ACD (3.00) (100 Agent Pkg)
CallCenterWorX-B ACD (3.00) (120 Agent Pkg)

CallCenter*WorX* ACD 3.0 for Business Software Only Packages

Business Software Only Packages
CallCenterWorX-B ACD (3.00) (10 Agent) SW Only
CallCenterWorX-B ACD (3.00) (20 Agent) SW Only
CallCenterWorX-B ACD (3.00) (30 Agent) SW Only
CallCenterWorX-B ACD (3.00) (40 Agent) SW Only
CallCenterWorX-B ACD (3.00) (60 Agent) SW Only
CallCenterWorX-B ACD (3.00) (80 Agent) SW Only
CallCenterWorX-B ACD (3.00) (100 Agent) SW Only
CallCenterWorX-B ACD (3.00) (120 Agent) SW Only

CallCenterWorX MIS 3.0

CallCenterWorX MIS 3.0 runs on the Windows XP and Windows 2000 operating systems. It provides MIS services for both CallCenterWorX – Business and CallCenterWorX Enterprise.

The CallCenterWorX ACD 3.0 for Business suite includes the Automatic Call Distribution (ACD) system and the Management Information System (MIS). The CallCenterWorX ACD system supplements the call processing capabilities of the platform being used. Typical call routing is designed to connect each caller with the most appropriate agent to handle the call in the shortest time possible, avoiding long wait times and getting the caller directly to someone who can deal with their request.

As each call progresses through the system, the ACD system provides call-handling data to the MIS, which translates this data into meaningful statistics. The MIS uses these statistics to offer call center managers and supervisors access to real-time and historical data concerning incoming and outgoing call volume in the call center.

The CallCenterWorX MIS Status Screens (Real-time Screens) provide color-coded, up-to-the-second views of agent, queue, and system activity. Statistics on the call volume, and on the effectiveness of the agents handling those calls, are computed in real-time and can be displayed up-to-the-second on supervisors' screens. Optional wall display boards can show current statistics and messages to call center personnel. The call statistics are also collected in the MIS database and used to generate reports in text or graphical format. These reports and graphs can be printed for views of activity ranging from a graph of today's activity to a retrospective report covering previous months.

The following features are available with the purchase of CallCenterWorX MIS 3.0.

- True Multi-Tenant Support
- Increased Multi-Split Agents (from 4 to 16 split monitoring)
- Microsoft MSDE
- ODBC Compliant
- Configurable in 15 min. / 30 min. / 1 hr Increments
- Conversion utility to migrate proprietary database to ODBC database
- Backup and Restore
- 1.5 GB database storage
- New "System Agent Detail Status" Screen
- Configurable Agent State Colors
- Retains Real-Time Screen Configuration

Business System Capacities

Item ACD	Capacities	
	CallCenter*WorX* ACD 3.0	CallCenter*WorX* MIS 3.0
Tenants per System	9	
Splits per ACD Tenant	**Note 1**	**Note 1**
Splits per System	150	150
Splits per Logon ID	32	32
Active ACD Agents per System	120 **Note 2/ Note 4**	120 **Note 2**
Trunks per System	255 **Note 4**	255
Maximum Queue Depth	300	–
Priority Levels 1	250	**Note 3**
Maximum Calls in Queue per System	300	–
Transfer to PBX Numbers	120	–
Agent Logon ID Codes per System	400	400
Agent Logon ID Code Digits	9	9
Statistics Report History Limit	–	**1.5GB**
MIS per System	8	8
Pilot Numbers	300 **Note 4**	300
Tally Code Names per Split	–	100
Tally Code Names per System	–	3000
Busy Hour Call Attempts (BHCA)	**Note 5**	–

Note 1: *The total number of splits per system (150) is shared among the total number of tenants (up to 9).*

Note 2: *Agent positions available depend on the installed security key's capacity for logged-on agents. The total number of agents is shared among the total number of tenants (up to 9). Security keys are available in the following capacities: 10, 20, 30, 40, 60, 80, 100, & 120.*

Note 3: *The MIS tracks Split queue calls as "High" and "Standard."*

Note 4: *Dependent upon the CallCenterWorX configuration formula based on the OAI monitoring limitations: (Agents x 2) + (Pilots and/or Personal Pilots) + Trunks + IVR Ports must be equal to or less than 256.*

Note 5: BHCA (Busy Hour Call Attempts)

Number of PIM's	1PIM	2PIM	3PIM	4PIM	5PIM	6PIM	7PIM	8PIM
Business or Hotel/Motel with ACD/OAI	Max. 2000 BHCA (No FP)		Max. 4000 BHCA (One FP)		Max. 5000 BHCA (Two FPs)		Max. 5500 BHCA (Three FPs)	

PBX Requirements

PBX Requirements	Remarks
PZ-M606-A	On board Ethernet Interface card (Ethernet TCP/IP)

Related Documents

Title
CallCenter*WorX* System Manual
CallCenter*WorX* Features and Specifications Manual
Dterm Series E Supervisor Console User Guide
Dterm Series E Agent Console User Guide
CallCenter*WorX* Sales Slick
ICS ACD System Manual
Abacus User's Guide
Navigator Configuration Guide
Navigator Reports Manual
Navigator Real Time Screen Manual
CallCenter*WorX* ACD User Guide

ACD Terminals

With the Dterm Series i/E ACD terminals enjoy the convenience of having the most frequently used ACD function keys silk-screened right on the terminal.

Availability

The 32-button Series E ACD terminals are available in soft white or black. User Guides and spare parts are also available.

DTP-32DA-1 (WH)
DTP-32DA-1 (BK)
Dterm Series E Supervisor User Guide (10-pack)
Dterm Series E Agent Console User Guide (10-pack)

Detachable dial-pads are available for Dterm Series I and Dterm IP instruments in an ACD environment.

BS(ACD)-R	Dial pad for ACD (Available in black and white)

Q-Master 3.1

Q-Master is a scalable contact center solution offering high-end functionality at a reasonable price. Q-Master is completely modular, allowing contact center management to select the components that best meet operational and budgetary requirements while retaining the flexibility to add additional modules in the future. One of the key advantages of this is the range and scalability of the solutions. This is the first contact center application available that is capable of providing the same functionality to end users regardless of whether they deploy a small, medium or large PBX or Key system using any of the *UNIVERGE* Family of systems or the Elite IPK. This creates a powerful migration path never before available to our customers.

Q-Master Contact Center Solutions

Contact Routing (Q-Control)

Q-Control is the core module of Q-Master. It lets users maximize contact center efficiency by providing dynamic control over the delivery of phone, email and web initiated contacts. Since labor is a contact center's largest expense, this efficiency equates to significant operational cost savings. Q-Control uses skills-based routing to match agent knowledge to the needs of different callers or groups of callers; agents handle the inquiries they are most equipped to deal with. When all the agents in a specific queue are busy, Q-Control automatically overflows calls to the next most suitable agent in another queue, by matching agent skill sets to a variety of different queues using a predetermined call delivery matrix. List of features included in Contact Routing (Q-Control):

- **Individual agent IDs** - Agents log on and off from their telephone or PC with a personalized ID number - the agent, not the extension is tracked.
- **Wrap-up codes** - prompts agents to supply a Wrap-up code, providing detailed real-time call handling information.
- **Work-time allowance** - provides each agent with post-call processing time.
- **Inter-Queue Transfer** - even when a call is transferred from one queue to another, callers maintain their position in the queue based on total time waited.
- **Wallboard Interface** - access real-time stats displayed on an electronic wallboard. A quick glance delivers an accurate picture of current performance levels.
- **Specialized features for supervisors** - a number of specialist features are also available for supervisors including:
    - on-demand call delivery
    - remote log in/out option
    - conversation monitoring/intrusion
    - emergency queue re-routing

- **Simplified Administration** – The wizards and a graphical user interface help to administer all aspects of the system. Adjustments to call delivery patterns, queue setup, agent assignment and other parameters can be made in a matter of seconds as updates are performed in real-time.

Reports (Q-Control)

Because access to timely and relevant information is critical for effective contact center management, a comprehensive reports package comes standard with every Q-Master solution, ensuring that users have the tools to run a contact center at peak efficiency. An onboard database holds statistics on every facet of every contact from the time it arrives at the telephony switch through to its termination (cradle to grave), allowing managers to report on areas such as volume, agent activity, abandonment rates, wrap-up codes and service levels. More than 120 reports are available detailing all aspects of the contact center operation. List of feature included with Reports (Q-Control):

- Detailed queue and agent reporting analysis
- Business reports on wrap-ups and service levels
- Ability to reformat data into graphs and tables
- Single integrated reports package covering telephony, email, web-based and outbound contacts
- Automatic report scheduling Simplified Wizards to configure and run reports data export to compatible packages
- Historical data archiving and retrieval

Announcements (Q-Announce)

Holding for any length of time is a frustrating experience for callers and projects the wrong image for a business. With Q-Announce users can configure each queue to play informative, specific messages that a hold a caller's attention and reduce the chance that they will abandon the call. With Calling Line ID enabled Q-Announce can recognize and play customized messages to a particular customer. Perhaps the most powerful feature of Q-Announce is its ability to keep callers informed of their position in the queue and the estimated wait time. This improves customer satisfaction and ensures that callers won't hang up just before they reach the front of the queue. List of features included with Announcements (Q-Announce):

- **Progress Announcements** - this simple technique significantly reduces the number of abandoned calls. Our results have proved that informed callers are 30% more likely to wait on hold.
- **Automated Attendant** - splits a single telephone number into numerous queues. Callers are then prompted to select the appropriate destination from a list, reducing wait times and connecting customers to the correct agents.
- **Multi-language Capability** – record each queue announcement in multiple languages. The language played to the caller is determined by the inbound number they dialed, the menu option they selected or their calling line ID.
- **Audiotext** - pre-recorded information menus provide answers to commonly asked questions. Audiotext saves valuable agent time that could be more effectively spent handling callers with complex queries.
- **Customer Query** - identify callers based on their calling line ID or their response to an automated query (such as "please enter your customer number") and use this information to play customized messages, direct the call to a specific agent or escalate their priority within the queue.

Agent Desktop (Q-Desktop)

As the importance of contact centers increases, so does the need to monitor operational parameters. Q-Desktop lets managers and agents view real-time information on queue and agent performance across all contact media, whether phone, email or the web, providing supervisors and agents with the tools for making the right decisions at the right time.
Agents have access to a wide range of contact center functionality directly from their PC desktop, ensuring that they can operate at optimal efficiency. Using the Agent toolbar they can:

- Log into and out of queues
- Request post call work time
- Take a break
- Wrap up or resolve a call
- Raise an alert for assistance

Q-Desktop also gives agents control over phone functions (e.g., answer, transfer, conference and hold) from their PC and identifies incoming calls by screen popping prudent information such as ANI, wait time, queue name, as well as caller's name account number and any other desired information from Q-Master or external database. Any actions taken by the agent in the contact center are reflected in real-time by Q-Desktop. These views give agents and supervisors a lot of valuable information at a glance.
List of features included with Agent Desktop (Q-Desktop):

- **Real-time information** - view the status of each agent and queue. Supervisors can monitor contact center service levels and act immediately when issues arise unexpectedly. Experience has shown that agent commitment increases when staff can view each other's performance. Your agents become increasingly goal oriented and require less supervision, resulting in increased productivity and accountability**.**
- **Caller information display**- Q-Desktop screen-pops calls as the phone or email is delivered to the agent. Using Caller ID, users can view the caller's name and number before answering the call, saving valuable seconds. Use the Phonebook, Q-Desktop's own database, to store special details for a caller, which may be screen-popped with the call.
- **Multiple views** – users can make their view of the contact center as simple or as detailed as they wish, and flick from one view to another with a single mouse-click. They may choose to view:
    - the queues and which agents are logged into
    - the individual agents and the type of call they are on
    - both queues and agents
- **Precise and convenient dialing** - point-and-click directories ensure fast and accurate dialing. Agents can also dial directly from their PC keyboards.
- **Specialized features for supervisors** - supervisors will always have a current picture of contact center activity: queue status, service level, abandonment rates, traffic volumes and agent statistics. A number of specialist features are also available for supervisors, including:
    - On-demand call delivery
    - Remote log in/out option to control agent status
    - Conversation monitoring and intrusion
    - Direct access to Q-Master's administration package - for dynamic changes
    - Emergency queue re-routing
    - At-a-glance icon indications of call and agent status

Callbacks (Q-Callback)

Callback allows callers who are waiting in the queue to leave a message and then hang up. The callback request retains its position in the queue, so when an agent becomes available the caller is automatically contacted as if they'd stayed on the line.

- **Gives Your Callers Control** - while some callers are content to wait on the line, many are not. Q-Callback invites callers to enter their contact number, leave a message and hang up, without losing their position in the queue. Callers using the Callback feature are handled just as quickly as if they had waited on the line.
- **Lower abandonment rates** - waiting on hold can be frustrating, costly and time consuming for your customer. You may also pay the price for leaving your caller on hold for "too long," causing them to hang up and take their business elsewhere.
- **Makes It Simple for Agents** - Callback is very flexible and can be configured on a per queue basis. Use the Callback schedule to determine times of day that you want Callback to be available, and specify thresholds when it should apply. Agents and Supervisors can see the number of callback requests in the queue and make the adjustments to operating parameters as required.
- **Lower your network costs** - benefit from substantial telecommunications cost savings, as Q-Callback cuts the number of calls holding on your free phone numbers.
- **Answers Your Callers 24 Hours/Day** - Q-Callback can act as an automated answering service, for instance, after-hours. When agents log in first thing in the morning, calls from the night before are immediately presented to their workstation for handling. No voice mailbox needs to be cleared and no calls have to be requested.

Multi-Media (Q-Email, Q-Chat, Q-Fax)

Use Q-Master's multimedia modules to manage these communication media within your existing contact center infrastructure - in much the same way as you currently manage your telephony calls. Just as phone calls, emails and web interactions are also delivered to agents via a queue as soon as they become available. Contacts are fairly distributed among agents and monitoring and reporting on agent performance and response time is available across all media. These modules are fully integrated into Q-Master so individual agents can receive requests initiated via telephone, web or email. The properties of intelligent call delivery and skills-based routing are applied when distributing any type of contact to agents. List of feature included with Multi-Media (Q-Email, Q-Chat, and Q-Fax):

- **Q-Email** - integrating email communication into a contact center environment provides customers with accurate, timely responses to their inquiries in a consistently professional manner. Monitoring and reporting of the incoming email traffic also provides a total picture of all inbound communications.
- **Q-Web Callback** - lets potential customers request a Callback from within a website. These requests are routed to the most suitable agent in your contact center. The agent receives a screen-pop with the details of the request. When they click the dial button, Q-Master automatically establishes the call.
- **Q-Web Chat** - offers web users real-time online interaction with a contact center via text-based chat sessions with agents. Agents have the ability to "push" links directly to online customers. The possibilities are endless and the impact on customer service is significant.

Outbound Dialing (Q-Outdial)

Why wait for your customers to call? Your customers can generate additional revenue, proactively manage customers and improve the productivity of every outbound customer contact by intelligently managing outbound call campaigns from within the existing Q-MasterEX solution.

- **Blend Inbound and Outbound Calls** - agents can be set up to work in dedicated outbound queues or take a mix of inbound and outbound calls. By sharing agent resources between inbound and outbound call activity, contact center managers can optimize agent productivity in periods of low inbound activity.
- **Flexible Call Delivery Options** - two types of call delivery modes are available; Power Dial and Preview Dial.
- **Information on Call Outcomes** - resolution and wrap-up codes allow agents to record the outcome of every call, indicating the status and commercial outcome of a call.
- **Build and Manage Campaigns Easily** - Q-Outdial is administered and managed using an easy-to-use administration interface.
- **Fully Integrated Solution** - Q-Outdial utilizes a common interface to manage and deliver both inbound and outbound calls. Agents and supervisors can perform all required call delivery tasks in this environment, minimizing operational complexity and training, and maximizing labor efficiency.
- **Reports** - allows the contact center manager to view campaign information and details of specific campaigns in an easy-to-read format that helps manage outbound campaigns.

Interactive Voice Response (Q-IVR)

Q-IVR is fully customizable, interactive voice response technology that allows callers to interact directly to a database through the key pad on their phone to place orders, manage their account, check the status of an existing order or have other routine questions answered. What is unique about Q-IVR is that it is fully integrated with the call center. List of features included with Interactive Voice Response (Q-IVR):

- **Reduce call traffic to agents** – enable customer to place orders, manage their account, check the status of their order and get answers for routine questions automatically thought the use of the IVR. By enabling this feature, agents can address more complex issues, better utilize agent time and reduce customer wait time throughout the organization by eliminating time consuming simple tasks that customers can now handle on their own.
- **Make it a 24 x 7 operation** – after normal business hours customers can still utilize the IVR capabilities, allowing automated services 24 hours, seven days a week.
- **Reduce call abandonment** - since agents are now handling the more difficult questions, call volume will most likely drop significantly, therefore wait times and call abandonment will decrease.

Que*WorX4.0*

Que*WorX*® *4.0* is a suite of customer service and productivity-oriented applications. QueWorX utilizes NEC's CTI link, "Infolink" to deliver advanced applications to contact centers

These advanced applications are tightly integrated and specially designed to work in conjunction with NEC's CallCenterWorX ACD applications for both *UNIVERGE* NEAX 2000 IPS and *UNIVERGE* NEAX 2400 IPX systems. QueWorX integrates with CallCenterWorX-Enterprise (I), & CallCenterWorX ACD 3.0 for Business & Enterprise. QueWorX provides the same features and functionality regardless of the ACD platform it accompanies. In addition, QueWorX may be installed on the same server as the CallCenterWorX ACD 3.0 for Business or Enterprise.

QueWorX 4.0 Licensed Applications

Automated Attendant
Immediate Callback
Scheduled Callback
Web (Internet) Callback
Estimated Time to Answer
Queue Depth
ANI/Are Code Routing
Account Code Routing
Screen Pop
Soft Phone

QueWorX 4.0 Features:

Modular Licensing
Multilingual Announcements
Global Repeat Verification
Customer Profile Records Database
Unlimited Multi-level Automated Attendants
Customizable Announcements
Customizable Reports

The above items are detailed below:

Auto Attendant

The Auto-Attendant provides basic call routing functions to the caller. The caller receives a message with up to ten call routing options and the caller presses a key to initiate a selection. The caller's selection is processed as a transfer to a destination. Additional menu choices are available for customers with the multi-level automated attendant. The caller may be prompted for Account Code or User ID, which will then be displayed on the agent's Dterm phone as the call arrives. Alternatively, the Account Code or User ID may appear in the Que*WorX* Agent Screen Pop. This allows the agent to begin accessing a customer's profile while greeting the caller.

Immediate Customer Callback

This feature allows callers, based on their estimated time to answer, to elect to receive a callback from an agent instead of waiting in queue. By entering their telephone number and recording a brief message, their call will remain in queue until an agent becomes available, even though they have hung up. The Que*WorX* system will automatically dial the number entered by the caller when the agent receives the call. Bonus: The contact center does not incur toll charges for the time call remains in queue awaiting a callback.

Scheduled Callbacks

This feature allows callers to determine when an agent calls them back. This feature brings a new level of flexibility to the immediate callback feature, allowing customers to determine interactions with the contact center based on their timetable.

Internet Initiated Callbacks

This feature allows customers to request a callback from an agent while browsing the organization's Web site. Requests for callback will be placed into queue as are voice calls. When agents become available, they will receive a screen pop with collected customer information and URL information.

Estimated Time to Answer (ETA)

Que*WorX* can provide a caller with information on the wait time expected before an agent answers the call. An announcement will only be provided if the actual ETA falls within the programmed parameter thresholds. Announcements can be provided in three variations stating a number of minutes and seconds (“approximately 2 minutes and 45 seconds”), only a number of minutes (“approximately 2 minutes”) or a maximum number of minutes (“less than 3 minutes”).

Queue Depth

Que*WorX* can provide an announcement indicating how many callers are ahead of a caller in the queue. Threshold settings are set within the system to determine when to offer this announcement to callers. For example the 2nd through 15th caller may hear their position within the queue, such as “The are5 callers ahead of you”. Any caller falling outside of this range would not hear the announcement.

ANI / Area Code Routing

This feature allows calls to be routed based on one or a combination of the following factors – ANI (Automatic Number Identification), Account Code, Area Code. This feature allows callers to be routed based on a match of any of these criteria. For example, callers may be prompted to enter their account number as they enter the system. Based on a match by the Que*WorX* system, their call may be redirected to a group of agents (or their personal agent) at a higher priority.

Account Code Routing

This feature allows the system to route a caller to a specific split or agent based on Account Code.

Screen Pop

Provides an application that is installed at each agent's desktop. Agents can log into Que*WorX* and the CallCenter*WorX* system via this application. Customer information provided in the Customer Profile Records Database can be displayed in the form of a 'pop-up' at each agent's desktop as calls are received. Additionally, the softphone capabilities of this application can be utilized to allow agents to enter break mode, ready mode, enter tally codes, etc. right from the computers.

Soft Phone

Using the agent Soft phone application from the agent client PC, the agent has the ability to:

- Answer the incoming Call
- Release (disconnect) the Call
- Set Break Mode
- Tally the Call
- Transfer the Call
- Conference the Call
- Place the Call on Hold
- Retrieve a Call on Hold

Multilingual Announcements

Up to seven (7) language sets can be configured and recorded in Que*WorX*. Each set is selected based on pilot number to accommodate multilingual applications. For example, a pilot may be defined to send calls to a group of agents who support Spanish-speaking customers.

Global Repeat Verification

This feature allows callers to verify their customer input, such as account number of callback number. This feature helps ensure the proper handling of customer calls.

Customer Profile Records Database

The profile contains information specific to each customer. This information is used to display pertinent customer information to agents via the Que*WorX* Agent screen pop application.

Unlimited Multi-Level Automated Attendant

Up to 10 menu options and unlimited levels are recorded and offered to customers. With this option, customers can provide more options for callers. Multiple levels allow callers to provide more detailed routing. For example, when the PBX option is chosen from the menu, callers may receive additional selection choices, such as "For NEAX 2000 press 1, for NEAX 2400, press 2", and so on up to 5 levels deep. Up to 10 distinct automated attendants can be configured to accommodate multiple customer menus. Calls can be routed via the automated attendant to selected departments or groups of agents.

Customized Announcements

Que*WorX* can be configured with up to 89 custom announcements. These announcements are recorded in Que*WorX* and invoke utilizing the Call Control Vectors of the CallCenter*WorX* application with the "IVR Announcement #" command.

Customizable Reports

The Reports can be customized using the default reports as templates. The modified reports can be saved for later use.

Professional Services for Que*WorX*

While QueWorX provides an advanced set of features, some customers will require applications specific to their contact center that are outside the scope of QueWorX. With hundreds of successful solutions deployed Professional Services has developed many custom solutions that allow us to provide Call Center, CTI and speech applications that meet the customer's business needs and demands. Requests for these applications can be forwarded to your NEC representative who will work with NEC Professional Services to help define the requirements of unique applications or you may contact Professional Services via ProServ@necunified.com.

Chapter 13 Voice over IP (VoIP)

VoIP Solutions

The *UNIVERGE* NEAX 2000 IPS (Internet Protocol Server) is a communication system that can convert voice signals into IP packets and transmit them through the same data communication line as IP network (Intranet). Using this system, both voice and data communication lines are integrated into one communication line, and communication cost can be reduced.

The following VoIP solutions are available with the *UNIVERGE* NEAX 2000 IPS system.

VoIP Solutions	
Extended Enterprise IP Solution	H.323 Connection
IP Station (Peer-to-Peer)	Fax over IP
CCIS Networking via IP	Modem over IP
Remote PIM over IP	

Extended Enterprise IP Solution

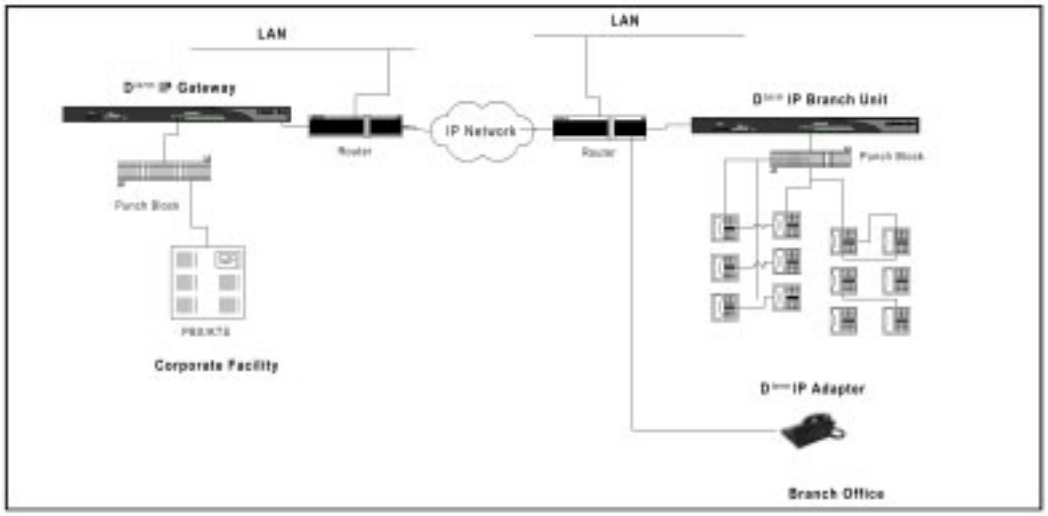
Teleworkers, usually working from an off-site location, can place a call through the corporate PBX/KTS by simply dialing the number as if they were in their office. The Dterm phones can be connected to either a multiple-user unit (called the Dterm IP Branch), or a single-user device (called the Dterm IP Adapter). The Dterm IP Gateway, Dterm IP branch and Dterm IP Adapter units are linked through a wide variety of network devices (TA, FRAD, and CSU/DSC) and network types (ISDN, IP, Frame Relay, T1, Fractional T1, ATM).

Dterm IP Gateway: is a Multiline module, available in 8 or 12 port units. On one side it has an AMP connector that is used to interface with Digital station ports. On the other side it connects to an IP packet network though a 10BaseT Ethernet port or circuit network through a DB25 WAN port. It communicates over the IP network to the Dterm IP Adapter(s) or over an IP network or circuit network to Dterm IP Branch unit(s).

Dterm IP Adapter: is a compact plug-and-play device that installs into the base of a Dterm display terminal. It has an integrated two-port 10BaseT Ethernet pass-though hub that permits using one port to connect the network interface card (NIC) from the PC to the IP network and the other port is plugged directly into a LAN or into an IP network device such as a router, DSL modem or cable modem.

Dterm IP Branch: Designed for small branch offices the Dterm IP Branch resides at the distant office. On one side it has an AMP connector that is used to interface with Dterm terminals. On the other side it connects to an IP packet network though a 10BaseT Ethernet port or circuit network though a DB25 WAN port. It communicates over the IP network or circuit network to the Dterm IP Gateway units.

Extended Enterprise IP Solution (Cont)



Typical IP Gateway, Branch and Adapter configurations

Required Equipment for NEC's Extended Enterprise IP Solution

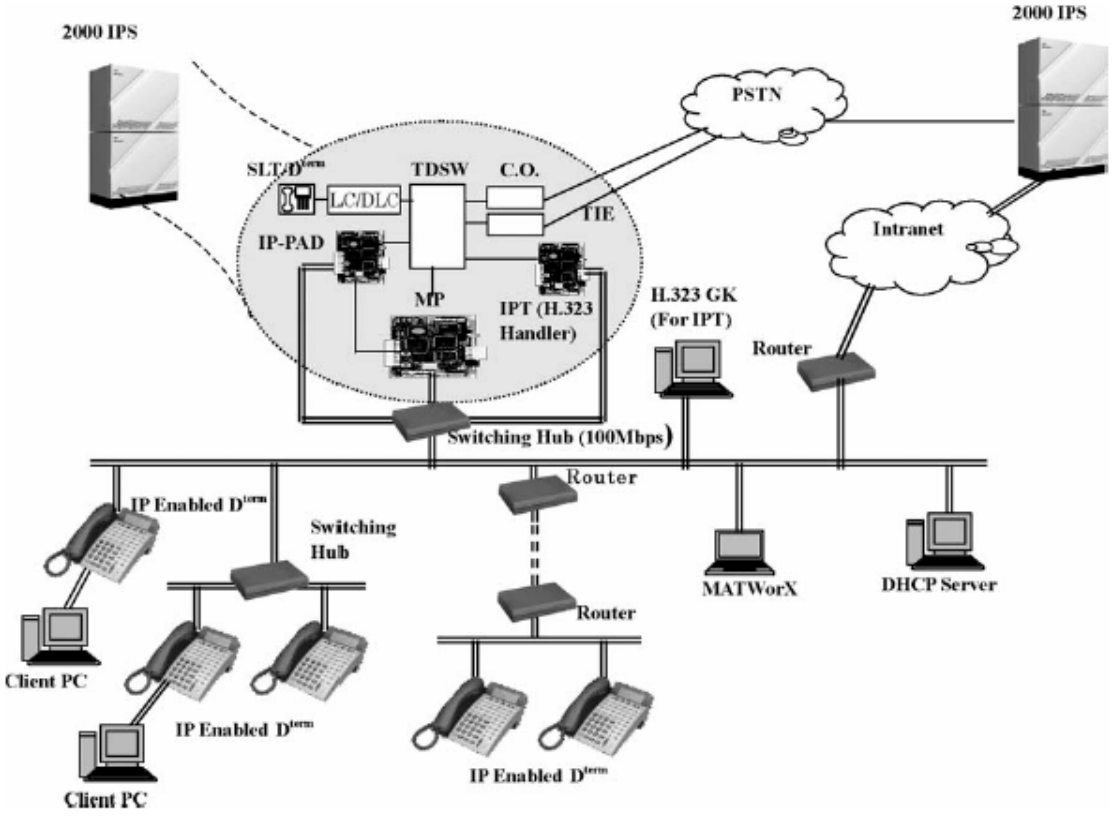
The table below shows the equipment list for NEC's Extended Enterprise IP Solution.

Equipment Name	Function
Dterm IP Gateway 12 Port Unit	The Dterm IP Gateway is a rack-mountable (or desktop) digital voice termination device that connects to the Dterm station ports of the NEAX or Elite system. The Dterm IP Gateway's main function is to extend Dterm station ports off of the corporate NEAX or Elite system to users over the LAN or WAN, such as telecommuters or small branch office workers.
Dterm IP Gateway 8 Port Unit	
Dterm IP Adapter (IPA-U10 Unit)	The Dterm IP Adapter connects to a Dterm Series E display terminal to provide voice access to a corporate NEAX or Elite system for a single user or local user. Voice and signaling traffic generated by the Dterm Series E will be placed in IP packets and sent out to the LAN via a 10Base T Ethernet connection to the Dterm IP Gateway. Used for connecting a single user.
Dterm IP Branch 12 Port Unit	The Dterm IP Branch converts NEAX or Elite systems voice and signaling into packets, which can be sent and received through a network connection. The Dterm IP Branch supports up to 8 or 12 users per stackable unit. Used for connecting multiple users from a single location.
Dterm IP Branch 8 Port Unit	

IP Station

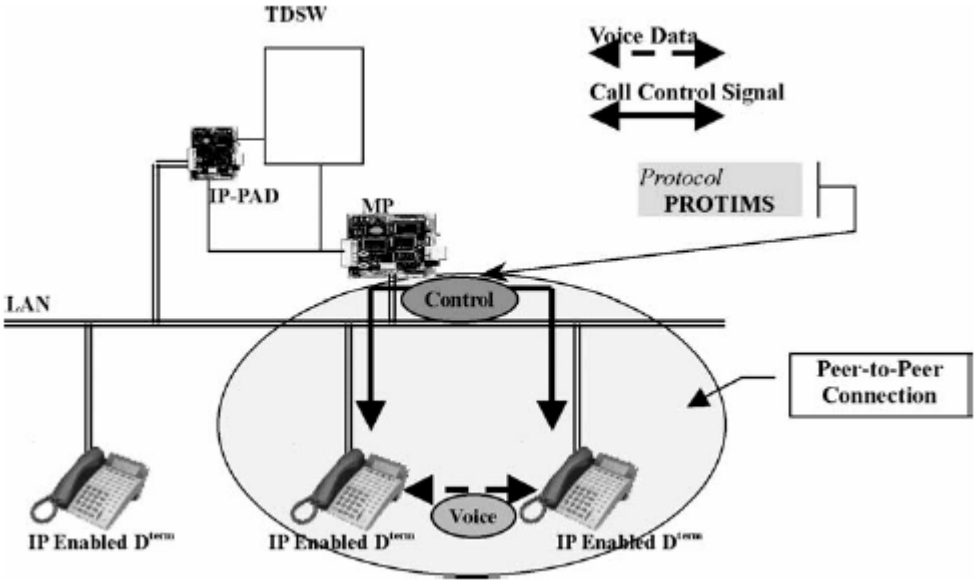
IP station is a DtermIP terminal, Dterm Series E/Series i with an IP Adapter Unit, INASET, and Dterm SP30/SP30 ACD Softphone. IP stations provide a converged infrastructure at the desktop, with a 10Base-T/100Base-TX Ethernet connection to corporate Local Area Networks (LAN). The IP station can communicate with other IP stations or CCIS network (IP based) on a peer-to-peer connection basis and the IP stations can communicate with legacy stations and trunks (TDM based) via IPPAD (IP Packet Assembler/Disassembler). The IP stations provide users with all features currently available in Dterm Series E/Series i terminals.

The following figure shows a typical network configuration using IP stations.



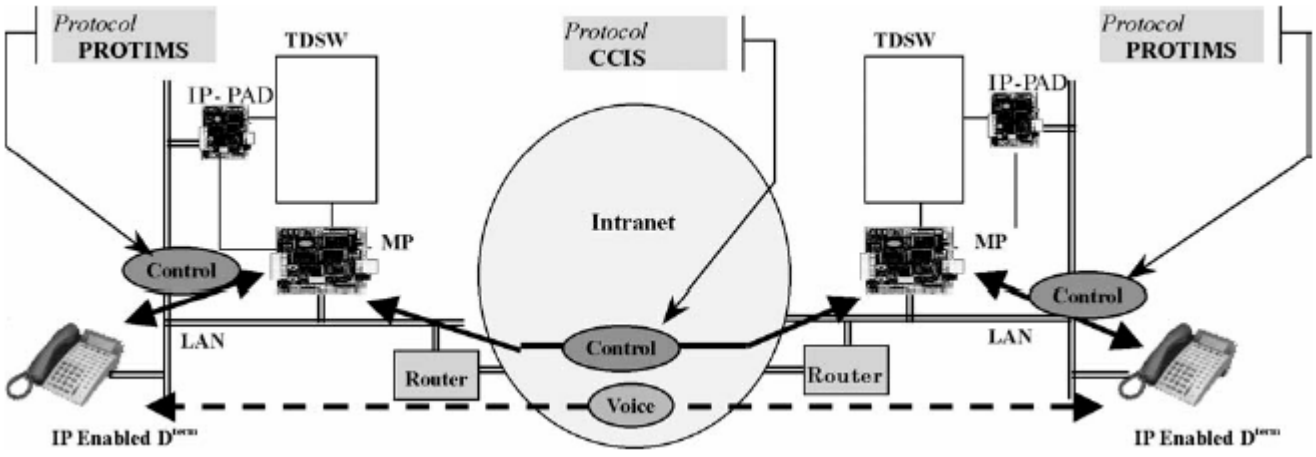
Peer-to Peer Connections between IP Enabled Dterm

The IP stations can communicate with other IP stations over the LAN, on a peer-to-peer connection basis. Call control is provided by the Main Processor (MP) with Ethernet adapter card, and voice packets are transmitted between IP stations over the LAN (not through Time Division Switch). Voice compression of G.729a (8 kbps) and G.723.1 (5.3 kbps/6.3 kbps) is available for its connections.



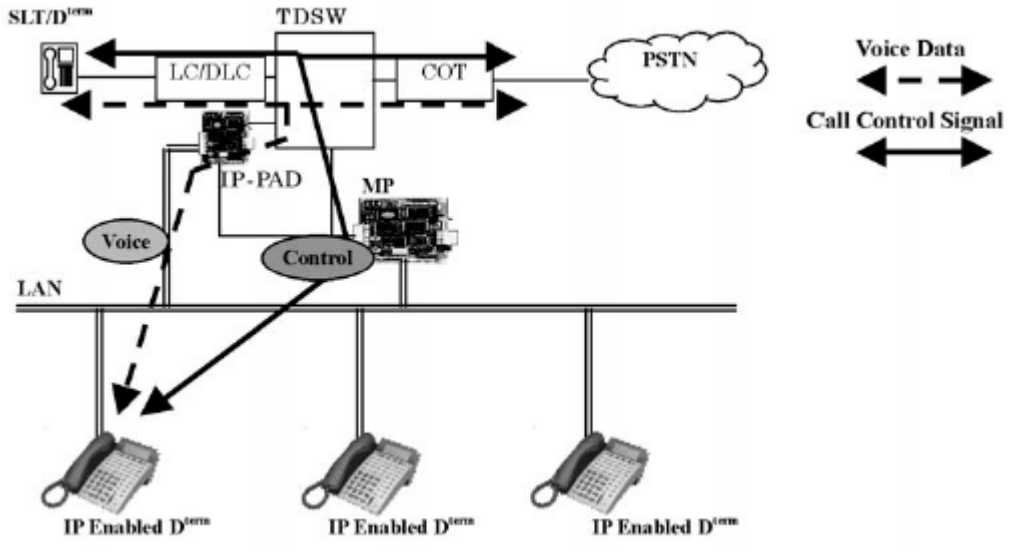
Peer-to-Peer Connections over CCIS Networking via IP

The IP stations can communicate with other 2000 IPS/2400 IPX systems over the CCIS networking via IP, on a peer-to-peer connection basis. MP controls a connection between the IP station and IP trunk built into the MP. The built-in IP trunk communicates with IP trunks of the distant systems (2000 IPS/2400 IPX) over CCIS networks via IP. Voice packets are transmitted over the LAN and WAN (via router). Voice compression of G.729a (8 kbps) and G.723.1 (5.3 kbps/6.3 kbps) is available for its connections.



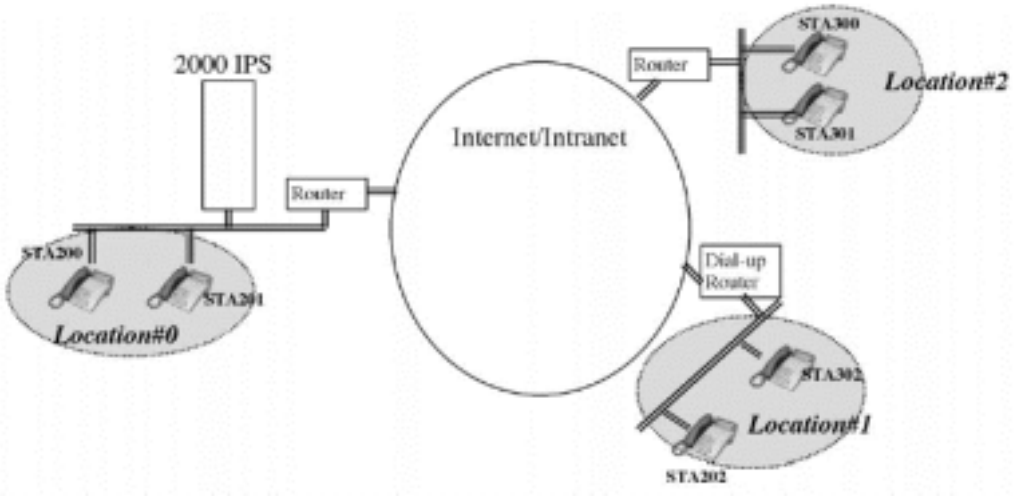
Connections to Legacy Stations and Trunks

The IP station can communicate with legacy stations and trunks (including H.323 trunks) via IP-PAD that converts voice packet data to PCM signals. Call control signals are transmitted to MP over the LAN, while voice packets are transmitted via IP-PAD. The number of IP-PADs depends on the traffic volume of connections between the IP station and legacy stations and trunks.



Remote Connections of IP Stations

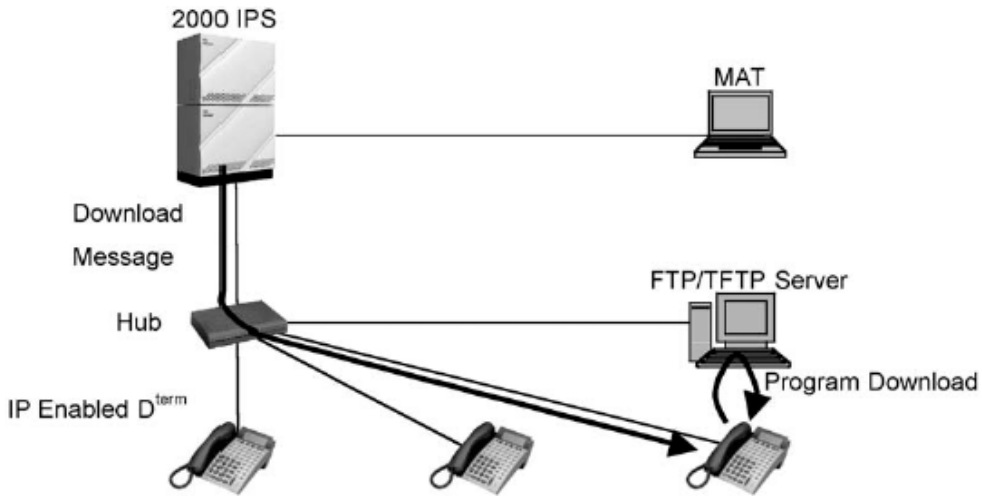
The IP stations can be located at remote sites over an intranet/internet/PSTN/ISDN via routers. IP stations can be installed behind routers that are configured to use NAT (Network Address Translation) at the remote location. This allows IP station users at remote sites to use the *UNIVERGE* NEAX 2000 IPS features as same as those at local site. The number portability among the sites is available, so that the same IP station user can login at any site, using the single login code and password. The IP station has two operating modes: Local Connection Mode and Remote Connection Mode. Each operating mode can have individual VoIP communication parameters such as CODEC selection list and usable bandwidth, so that the appropriate communication is available even if the IP station user logs in at either local site or remote site.



Automatic Program Download for IP Enabled Dterm

This feature provides the method to download the latest firmware program of IP Enabled Dterm from the FTP/TFTP server automatically by system programming. The following three patterns of program download are available.

- Program Download at Appointed Time
- Program Download at Login Time
- Program Download for Designated Terminals



Service Conditions

1. Up to 956 IP terminals can be accommodated per system.

2. Total number of ports for IP terminals, legacy stations/trunks (except for AP cards) and IP PADs is up to 1020 per system. See chapter 1 for capacities of stand alone and network configurations.
   Note: *Legacy stations/trunks means the ones connected to the Time Division Switch (TDSW). Examples of legacy station/trunk cards are LC, DLC, COT, PRT, CCT, DTI, 8RST (DTMF receivers), CFTB (Conference trunk), etc.*

3. IP terminals are accommodated in the Virtual PIM's (Virtual FPs) so that the hardware-PIM's are not required to accommodate IP Enabled Dterm terminals. The Virtual PIM is software-defined PIM used for data assignment for IP Enabled Dterm terminals and Virtual IP trunks (for CCIS Networking via IP). Maximum nine (15) Virtual PIM's are provided per system. Total number of the Virtual PIM's (for IP terminals) and hardware-PIM's is up to 16 per system.

4. Minimum one hardware-PIM is required per system to accommodate the MP card. The number of hardware- PIM's depends on the number of legacy stations/trunks and IP-PADs. Up to two hardware-PIM's (128 LT ports) can be controlled by the MP without the addition of an FP card. With three FP cards, up to eight hardware-PIM's (512 LT ports) can be controlled [One FP card controls up to two hardware- PIM's (128 LT ports)].

5. The IP-PAD is required for connections between IP terminals and legacy stations/trunks. The IP-PAD converts voice packet data to PCM signals. The number of IP-PADs depends on the traffic volume of those connections. Up to eight IP-PADs can be accommodated per system, thus providing 256 channels. Up to two IP-PADs can be controlled by the MP without FP cards, and additional IP-PADs can be controlled by the FP cards (two IP-PADs per FP card).

- 8IPLA/24IPLA
  PN-8IPLA card provides 8 channels of IP-PAD with VCT function per card. Adding PZ-24IPLA daughter-card on the 8IPLA card provides up to 32 channels of IP-PAD with VCT function. Maximum of 8 IP-PAD cards can be accommodated per system, thus providing 256 channels. By system data setting, the used ports for IP-PAD can be changed to 8/16/24/32 ports.
  Note: *Up to 24 channels of IP-PAD are provided when using G.723.1 codec.*

6. The following types of connections are available on a peer-to-peer basis:
   - Connections between IP terminals
   - Connections for CCIS networking via IP from/to IP terminals (2000 IPS or 2400 IPX)

7. The IP-PADs are required for the following connections/statuses:
   - Connections between IP terminals and legacy stations/trunks/IP trunks(H.323)
   - Connections for CCIS networking via IP from/to legacy stations/trunks
   - While IP terminals are on hold (Consultation Hold, Call Transfer, Music-on-Hold, etc.)
   - When any override service is activated (Executive Override, etc.) including IP terminals.
   - Three/four-party conference including IP terminals

8. Voice compression is available for following connections.
   - Connections between IP terminals
   - Connections via IP-PAD (for CCIS networking via IP from/to legacy stations/trunks)
   - Voice compression can be assigned on a call basis or terminal basis, by system programming.
   - The following type of voice compression is available:
     G.711 (64 kbps)
     G.729a (8 kbps)
     G.723.1 (5.3 kbps/6.3 kbps)

9. The Device Registration Server (DRS) is built into the MP card (System-based DRS). The System-based DRS can provide log-in/log-out function and registration authorization function of IP terminals. Up to 956 IP terminals can be managed by the System-based DRS.

10. Payload size of voice packet can be assigned from 10 milliseconds to 40 milliseconds in 10 milliseconds increments by system programming. However, available payload size depends on the type of voice codec.

Codec Type	Payload Size	
	Available Range	Default Value
G.711	10 milliseconds to 40 milliseconds	40 milliseconds
G.729a	10 milliseconds to 40 milliseconds	40 milliseconds
G.723.1	30 milliseconds (fixed)	30 milliseconds

11. Maximum number of voice channels per IP-PAD card are as follows:

- 8IPLA/24IPLA

Payload Size	Maximum Voice Channels per IP-PAD Card (8IPLA/24IPLA)		
	G.711	G.729a	G.723.1
10 ms	20	20	-
20 ms	32	32	-
30 ms	32	32	24
40 ms	32	32	-

12. Following parameters can be assigned on a Location basis. The Location is a kind of group (similar to Tenant in a PBX) that is defined for bandwidth management and parameter settings for LAN traffic based on the location of IP terminals. Maximum 64 Location numbers can be assigned per system.

Options Per Location		
Type of Codec selection list	DiffServ	Operation when over usable bandwidth threshold.
Type of Service (ToS)	Setting Area/Country	
PAD control pattern	IP-PAD Group number	Jitter buffer
Echo canceller-ON/OFF	Usable Bandwidth	

13. When the IP adapter unit is attached with a Dterm Series E/Series i terminal, other optional adapters (except for AD/DC adapter unit) cannot be attached with the Dterm Series E/Series i terminal.
14. Dynamic Host Configuration Protocol (DHCP) cannot be used to assign an IP address to an MP.
15. Login is restricted in following conditions.
    - When designating a non-existent station number. ("**Unregistered**" is displayed on the LCD.)
    - When designating a station number that is not assigned to Multiline Terminal. ("**Illegal LEN**" is displayed on the LCD.)
16. Logout should be allowed by a station class of service. If not allowed, reorder tone will be heard and "**Restrict**" will be displayed on the LCD.
17. Intranet to support Quality of Service (QoS) is preferable to connect IP Stations, to reduce the delay of connection and voice, and the inferior grade of voice quality.
18. IP Stations operate on 10 Mbps or 100 Mbps LAN based on IEEE 802. (100Mbps LAN is recommended.)
19. IP Stations support Dynamic Host Configuration Protocol (DHCP). (External DHCP server is required.)
20. Network-based DRS (external DRS) 3.0 is not supported.
21. Switching hub(s) are required.
22. Spanning Tree (IEEE 802.1d) function is not available.
23. LACP (Link Aggregation Control Protocol, IEEE 802.3ad) function is not available.
24. The *UNIVERGE* NEAX 2000 IPS can configure the speed mode of the LAN interface for MP/IP-PAD card, in addition to Auto Negotiation (default).
    - The speed mode for the MP card can be set to 100 Mbps full duplex in system data programming. System initialization is required for the mode to take effect.
    - The speed mode for the 32IPLA/32IPLA-A card can be set to 100 Mbps full duplex, 100 Mbps half duplex, or 10 Mbps half duplex, using the rotary switch on the card. The card initialization is required for the mode to take effect.
    - The speed mode for the 8IPLA card can be set to 100 Mbps full duplex using the DIP switch on the card. The card initialization is required for the mode to take effect.
25. When the QoS Display function key is pressed on IP Enabled Dterm, the QoS information (number of lost packets, codec type and payload size) is displayed.
    - A function key on IP Enabled Dterm should be assigned as QoS Display key in system data programming.
    - No other LCD information from PBX such as Clock/Calendar, calling station number and name is displayed during QoS display.
    - To stop QoS display, press the EXIT soft key (not EXIT button). Even if the call ends and the Dterm user goes on-hook, the QoS display remains unless pressing the EXIT soft key.

Service Conditions on Legacy Service Features

1. A DSS/BLF Console can be associated with the IP Enabled Dterm terminal, but the console must be connected to a DLC card.
2. Multiple Line Operation with normal Multiline Terminal is available.
3. Service features requiring continuous voice transmission, such as the Background Music feature, cannot be used because this traffic may reduce overall performance of the Local Area Network (LAN).
4. CAT mode is available.
5. Soft Key and Help indication is supported.
6. Set Relocations is not available between IP Station and TDM Multiline terminal.
7. DTMF signal is sent for 112ms ~ 128ms when Key Pad is pressed.
9. IP terminals do not support Add-on Module feature.
10. IP Enabled Dterm uses only one synthesized melody (Minuet), built into an IP adapter unit. Music on Hold using external source is not available for the IP Enabled Dterm.
11. Internal Zone Page from the IP terminals has the following conditions:
    - IP Enabled Dterm can page up to 3 Dterm/IP Enabled Dterm terminals.
    - Legacy Dterm can page up to 16 Dterm/IP Enabled Dterm terminals.
    - IP stations cannot provide all zone paging (receive reorder tone).

Service Conditions on System Registration

1. The IP Station can be registered in the *UNIVERGE* NEAX 2000 IPS in Login method (with password protected) or Automatic Login method (MAC address authentication). The registration method can be assigned in a station class of service.
   Note: *the DtermSP20/SP30 (soft phone) is registered in Automatic Login method only.*
2. The Login method allows the IP station user to be registered in the system by entering its own login code (station number) and password. The station user can login to the system from any IP Enabled Dterm terminal in the system, which is assigned to Login method. The Automatic Login method (MAC address authentication) allows the IP Enabled Dterm to be registered in the system at the installation time by entering its own login code (station number) and a special password for installation engineers.

   Once the terminal is registered, the station user does not have to login/logout to use the IP Station.
3. Up to 8-digit password can be assigned by system programming. 0 to 9, A and B can be used as a password. The password can be masked by "*" on the LCD of IP Enabled Dterm.
4. The registration data by Automatic Login method (MAC address authentication) can be backed up in the flash ROM of the MP by MAT operation or automatic system data back up at a designated time. Therefore, the installation engineer does not have to re-register the IP Enabled Dterm when the system reset should occur. The registration data by Login method (with password protected) cannot be backed up.

5. When the IP terminal with a call in progress has the LAN cable extracted or a power-off occurs and restores, in this situation, **"Double Assignment"** is displayed on the LCD. When the terminal is registered by Automatic Login method the terminal can be used automatically after about two minutes. When the terminal is registered by Login method the station user can login from the same terminal after about two minutes, or login from another terminal (Override).

6. Logout operation from the IP terminal registered in Automatic Login method should not be done in the normal operation. If the logout occurs, re-registration is required.

7. The registration data by Automatic Login method can be saved from the MAT by designating the area number 80 (normal system data saving operation does not save the registration data).

Service Conditions on Encryption in System Registration

1. When the IP Enabled Dterm is registered in the *UNIVERGE* NEAX 2000 IPS, the login code (station number) and password entered from the terminal can be encrypted.

   Login code: Proprietary algorithm
   Password: Proprietary algorithm or MD5 algorithm
   MD5 is an algorithm defined in RFC 1321 from the IETF.
   Encryption algorithm can be assigned on a system-wide basis (no encryption is also available).

2. Encryption is available for Login method (password protected) and Automatic Login method (MAC address authentication).

3. Up to 8-digit password can be assigned. 0-9, A and B can be used as a password.

4. When the password is not assigned, the registration is not allowed and **"Unauthorized"** is displayed on the terminal.

5. When the encrypted password is manipulated in the network, the registration is not allowed and **"Unregistered"** is displayed on the terminal.

Service Conditions on VLAN

1. *UNIVERGE* NEAX 2000 IPS supports VLAN based on IEEE 802.1Q (Tag VLAN).

2. VLAN can be assigned to IP port(s) of MP, IP-PAD and IP Enabled Dterm, and cannot be assigned to IP ports of IP trunks and IPELC. The VLAN assignment to the IP Enabled Dterm is provided by Configuration Mode.

3. One IP port can have one VLAN ID. Multiple VLAN IDs cannot be assigned to the same IP port.

4. VLAN ID can be assigned from 1 to 4094. VLAN ID = 0 is handled as Null VLAN ID and is effective to assign priority only.

5. Switching hub must support VLAN. If the switching hub does not support VLAN, the VLAN function is not effective even when the system data of the *UNIVERGE* NEAX 2000 IPS is assigned.

Service Conditions on Remote Connections

1. The network between NEAX 2000 IPS and IP Enabled Dterm must meet the following requirement:
   - Waiting time for ACK signal: maximum 600ms (300ms recommended)
     To check the above requirement, send Ping command from a PC to a remote IP Enabled Dterm and see the result of the command ("Time" shows the time to receive ACK signal.). If the result does not meet the above requirement, abnormal operation such as connection unavailable, deterioration of voice quality or abnormal state of IP terminal may occur.

2. The remote IP Enabled Dterm can be connected to the network via either of the following devices:
   - Router
   - Dial-up router
   - Modem
   - Cable Modem
   - ADSL Splitter

3. The IP terminal can be assigned to Local Connection Mode or Remote Connection Mode by entering the Administrator Mode of the IP terminal. After the operation mode of the terminal is changed (Local to Remote, or vice versa), login operation is required again.

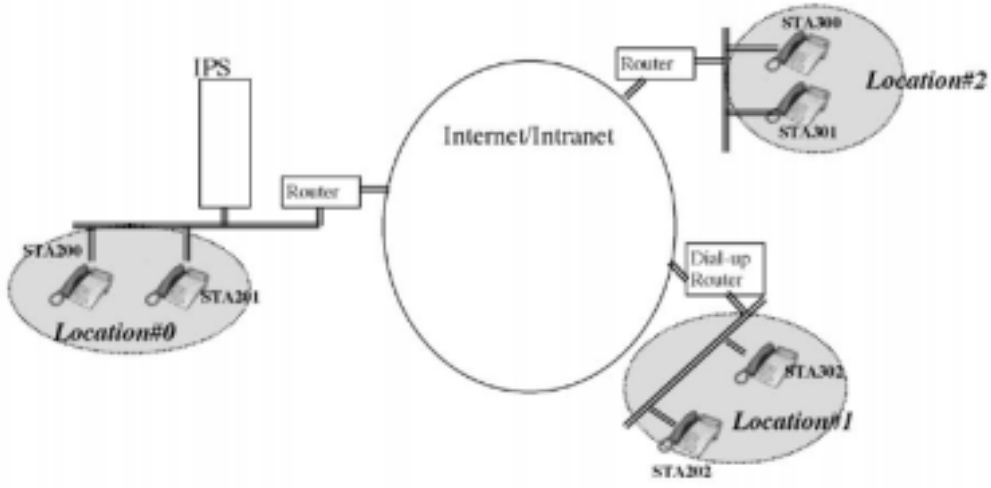
4. The differences between Local Connection Mode and Remote Connection Mode are as follows:

Item	Local Connection Mode	Remote Connection Mode
Confirmation to Connect IP Terminal	1 time per minute	1 time per day (10 minutes from 00:00 a.m.)
Clock Control	2000 IPS sends clock signal to IP terminal on a minute basis.	2000 IPS sends clock signal to IP Enabled Dterm when the IP Enabled Dterm is logged in or the clock data is entered by CM02.

5. Individual Location number can be assigned for Local Connection Mode and Remote Connection Mode, on a station number basis, by system programming. The IP Enabled Dterm can work based on the location number in each mode. When the Location number for the station is changed, login operation is required again.

6. NAT (Network Address Translation) is supported using remote Dterm IP terminals only. For NAT support a Global IP Address programmed in the NEAX 2000 IPS CPU is required.

7. Because the IP station can have only two-operation mode (Location number), when the number of remote sites (Location numbers) is over two, there is a case that the appropriate communications may not be available. Refer to Example 2 for details.
   - **Location number in each operation mode for each IP station:**
     <IP stations in Location #0: STA. 200 and 201>
     Remote Connection Mode: Location #1
     Local Connection Mode: Location #0
     <IP stations in Location #1: STA. 202 and 302>
     Remote Connection Mode: Location #1
     Local Connection Mode: Location #0 or #2
     <IP stations in Location #2: STA. 300 and 301>
     Remote Connection Mode: Location #1
     Local Connection Mode: Location #2
   - **Operation mode for each IP terminal:**
     <IP terminals in Location #0>
     Local Connection Mode
     <IP terminals in Location #1>
     Remote Connection Mode
     <IP terminals in Location #2>
     Local Connection Mode
   - **Example 1: When Sta. 200 logs in Location #1**
     STA. 200 logs out from Location #0, and then logs in from IP terminals in Location #1. STA 200 works in Remote Connection Mode (because IP terminals in Location # 1 are assigned to Remote Connection Mode.), under the conditions of Location #1 (because STA. 200 in Remote Connection Mode is assigned to operate under the condition of Location #1.). In this case, the appropriate communication is available because the operating condition of STA. 200 matches the actual Location number (Location #1).
   - **Example 2: When Sta. 200 logs in Location #2**
     STA. 200 logs out from Location #0, and then logs in from IP terminals in Location #2. STA. 200 works in Local Connection Mode (because IP terminals in Location # 2 are assigned to Locale Connection Mode.), under the conditions of Location #0 (because STA. 200 in Local Connection Mode is assigned to operate under the condition of Location #0.). In this case, the appropriate communication may not be available because the operating condition of STA. 200 (Location #0) does not match the actual Location number (Location #2).



8. Following features are not available for IP terminals at remote sites:
   - Multiple Line Operation
   - Assignment of feature keys which the lamp indication and ringer is controlled by other IP terminals

Service Conditions on Automatic Program Download

1. For program download, the FTP or TFTP server is required.

2. This service is available for the IP Enabled Dterm with Peer-to-Peer connections. Terminals with IPELC connections are not supported.

3. Maximum of four terminals can be downloaded at the same time regardless of the type of download.

4. Firmware version filed in FTP/TFTP server is assigned by the system data (CM0C YY=00-07, 1st=02). When the latest firmware version is filed in the server, be sure to update the system data.

5. When the FTP server is used for program download, login from the IP Enabled Dterm with a login name "**anonymous**" and password "**2000 IPS**" (authentication by FTP server is required).

6. Conditions on Program Download at Appointed Time
   - Program Download at Appointed Time service is available by the system data assignment (CM15YYY=482) on a station class of service basis.
   - By the same command mentioned above, program download can be retried just one time. When the download results in failure two times in succession, the IP Enabled Dterm operates with previous firmware.
   - When this service is executed, neither Program Download at Login Time nor Program Download for Designated Terminal can be executed.
   - PBX system checks the firmware version of each terminal starting from the terminal with the smallest LEN number and executes this service if required.
   - This service is executed only for the idle terminals. This service is not executed for the terminals in Logout or busy status.
   - The actual starting time of the Program Download at Appointed Time may be varied with a range of +/--1 minute. When 448 of IP Enabled Dterm have to be upgraded, for example, it takes about 450 minutes to be completed, without retry.
   - CMFA YY=50 1st=01/02 counts the number of terminals that are successfully downloaded and the number of terminals that result in download failure only when the PBX sends the latest program download message to the terminals. If the terminals are busy or logged out, CMFA YY=50 1st=02 does not count the number of terminals in download failure because the PBX does not send the download message to the terminals.
   - Program Download at Appointed Time can be suspended by CMFA YY=50 1st =00: CCC (clear) during executing the download. However, you cannot suspend the download of the terminal with "Downloading..." displayed.

7. Conditions on Program Download at Login Time
   - Program Download at Login Time service can be assigned on a system basis by CM0C YY=90 1st =02.
   - When this service is assigned, retry in download failure status is not available.

8. Conditions on Program Download for Designated Terminals
   - Program Download for Designated Terminals is assigned and executed on the Primary Extension number basis.
   - This service is executed only for idle terminals. This service is not executed for the terminals in Logout status.
   - When this service is assigned, retry in download failure status is not available.

Required Equipment for IP Station

Equipment Name	Function
PZ-M606-A	On board Ethernet Interface card. One per system
SPN-32IPLAA IP PAD	32ch IP Interface card for IP PAD
SPN-16VCTAA IP PAD	16ch Codec card for IP PAD Compression Max. 2 per 32IPLA Card (optional)
SPN-8IPLA IP PAD	8ch IP PAD card w/compression
PZ-24IPLA	24ch Expansion card for IP PAD. Mounts on 8IPLA.
IPW-2U Unit	IP Adapter Unit for Dterm Series E terminals
IP-R Unit	IP Adapter Unit for Dterm Series i terminals
AC-R Unit	AC/DC Adapter for Dterm Series i terminals
ACA-U Unit	AC/DC Adapter for Dterm Series E terminals
IP Terminals	DtermIP terminals Dterm Series i/E terminals w/IP adapter INASET terminals SP30/SP30 ACD Softphones
Power	For Powering the IP stations use one of the following: AC power supply (ACA-U) SN1604 Power Hub Cisco Catalyst switch power feed

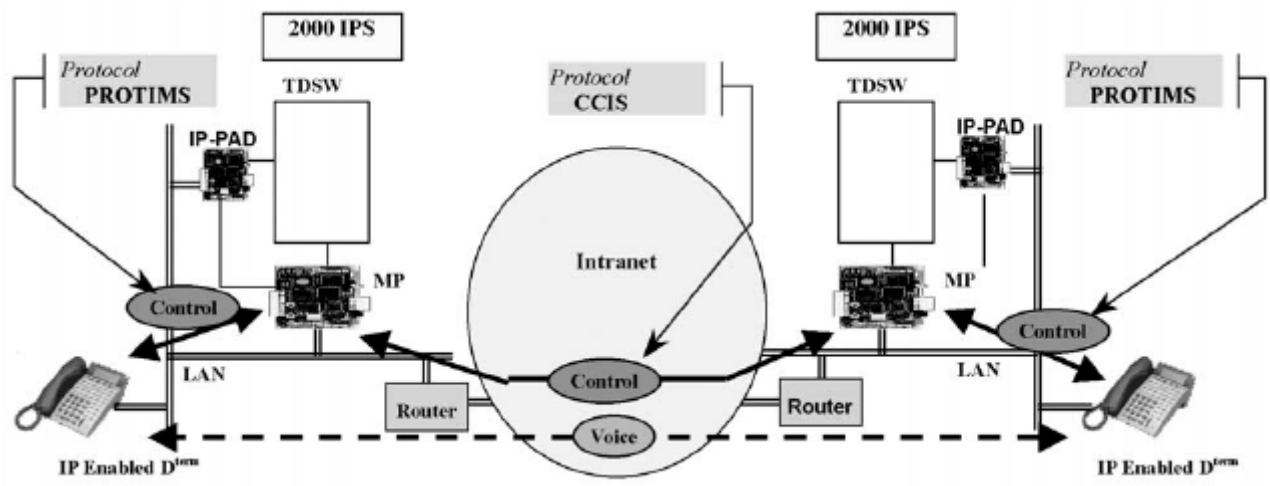
CCIS Networking via IP

This feature provides CCIS networks with Voice over IP (VoIP) capabilities to provide a converged infrastructure over corporate Wide Area Networks (WAN). The IP Enabled Dterm can communicate with other IP Enabled Dterm over the CCIS network (IP based) on a peer-to-peer connection basis. The legacy terminals (TDM based terminals) can communicate with other legacy terminals or IP Enabled Dterm terminals over the CCIS network via IP Packet Assembler/Disassembler (IP-PAD). Voice compression of G.729a (8 kbps) and G.723.1 (5.3 kbps / 6.3 kbps) is available for those connections. The CCIS Networking via IP provides users with all TDM-based CCIS functionality, such as feature transparency, centralized management, and centralized facilities.

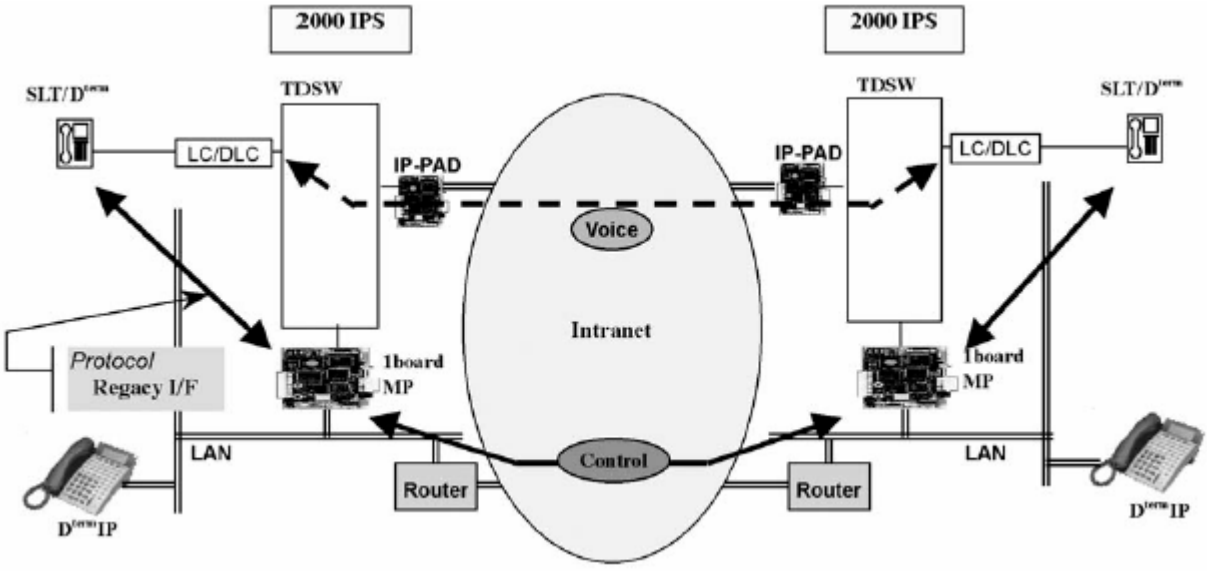
There are two types of connections available for CCIS Networking via IP:

CCIS Networking via IP (Peer-to-Peer Connections Basis)

When the distant systems are 2000 IPS, the systems are connected on a peer-to-peer basis. The CCIS call control signals are transmitted between the built-in IP trunks (CCIS Handler; CCH) on the MP card, over the Local Area Networks and Wide Area Networks (LAN and WAN). For connections between IP Enabled Dterm terminals, voice signals are also transmitted over the LAN and WAN. For connections between legacy terminals, voice signals are transmitted via IP-PADs. This connection is also available when the distant systems are 2400 IPX that supports peer-to-peer connections.



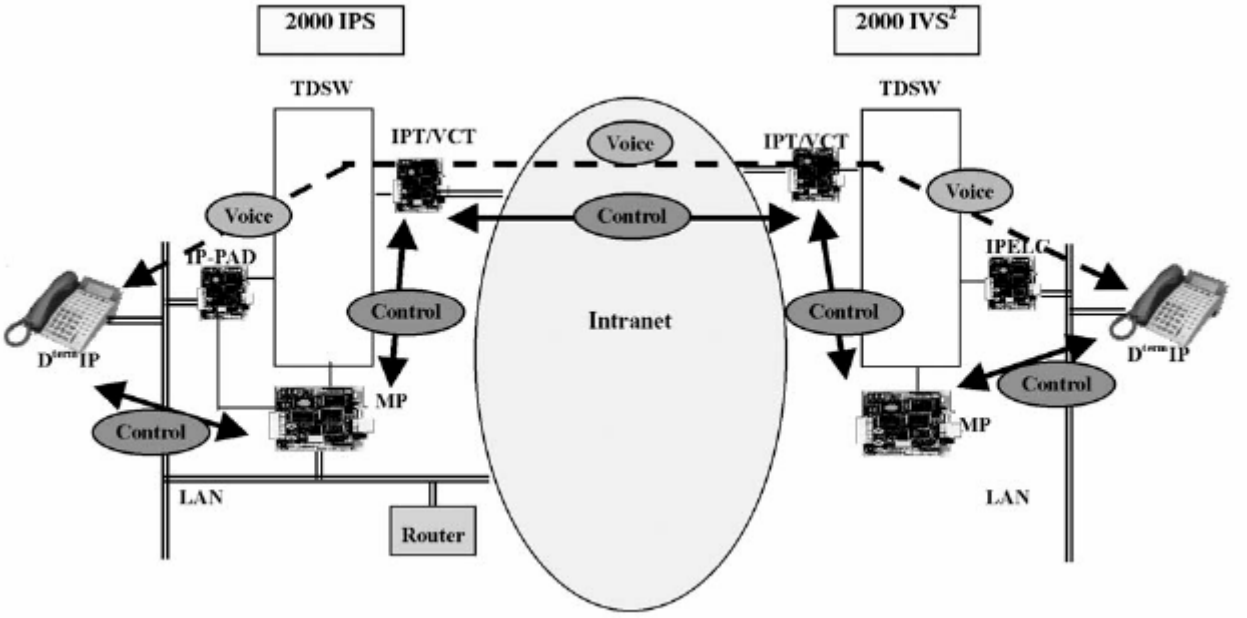
Example IP Enabled Terminals Peer-to-Peer Connection



Example IP between Legacy Terminals via Peer-to-Peer Connection

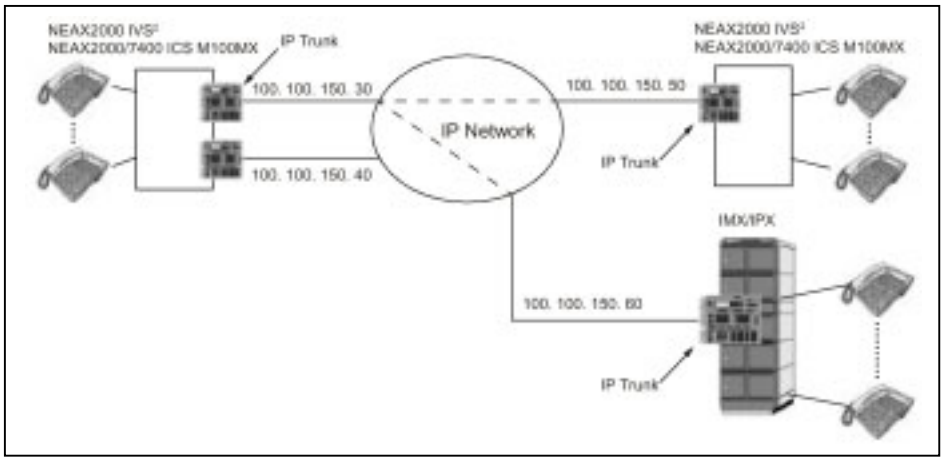
CCIS Networking via IP (Non Peer-to-Peer Connections Basis)

IP trunk connections over CCIS (No.7 CCIS Common Channel Inter-Office Signaling) Networking via IP is used to provide telephony services between the 2000 IPS, IVS^2, IPS Retro or 2400 IMX/IPX and Electra Elite IPK/IPKII on opposite side which do not support peer-to-peer connections. The IP-PAD is required for connections between IP terminals and IP trunks. A maximum of one IP trunk card can be accommodated per PIM with a maximum of eight IP trunk cards per system. Voice compression of G.729a (8 kbps) and G.723.1 (5.3 kbps / 6.3 kbps) can be provided by the IP trunks with VCT cards. Communications cost can be reduced, by utilizing IP Trunks in the system to convert voice signals into IP packets and transmit them over the data network. All CCIS centralized and transparent features can be used connecting up to 255 NEAX systems in a network for a total of 130,560 station ports.



Point-to-Multipoint Connection

Point-to-Multipoint Connection is a connection type, which designates multiple destination IP addresses, per IP trunk, as shown below. One IP trunk can be connected to multiple opposite offices in Point-to-Multipoint connection.



IP Trunk: Point-to-Multipoint Connection

Service Conditions (IP CCIS via IP Trunk cards)

1. Maximum voice channels per IP trunk card depends on the payload period as follows (payload period can be assigned in system programming):

Payload Period	Maximum Voice Channels per IP Trunk
10 ms	4
20 ms	8
30 ms	16
40 ms	16

2. Maximum 127 voice channels can be provided per system (point-to-multipoint connection).
3. IP trunk cards must be connected with the intranet via router.

IP Trunk Card Specifications

Description	Specifications
QoS signaling	IP Precedence/DiffServ
Speech encoding/decoding	G.711/G.729a/G.723.1
Payload period	G.711/G.729a : 20 ms, 30 ms, 40 ms G.723.1 : 30 ms (fixed)
FAX communications protocol	T.30
FAX mode detection timer	0 to 5 min. (1 min. increment)
PAD control	Controlled in a node basis
Tandem connection	IPT to IPT/IPT to ACIS: available (but voice quality may deteriorate) IPT to ISDN : available
IPT connection mode	Fixed Connection Mode or Automatic Disconnect Mode can be assigned per IPT trunk or destination node in system programming.
Dynamic jitter buffer	10 to 600 ms (10 ms increment)

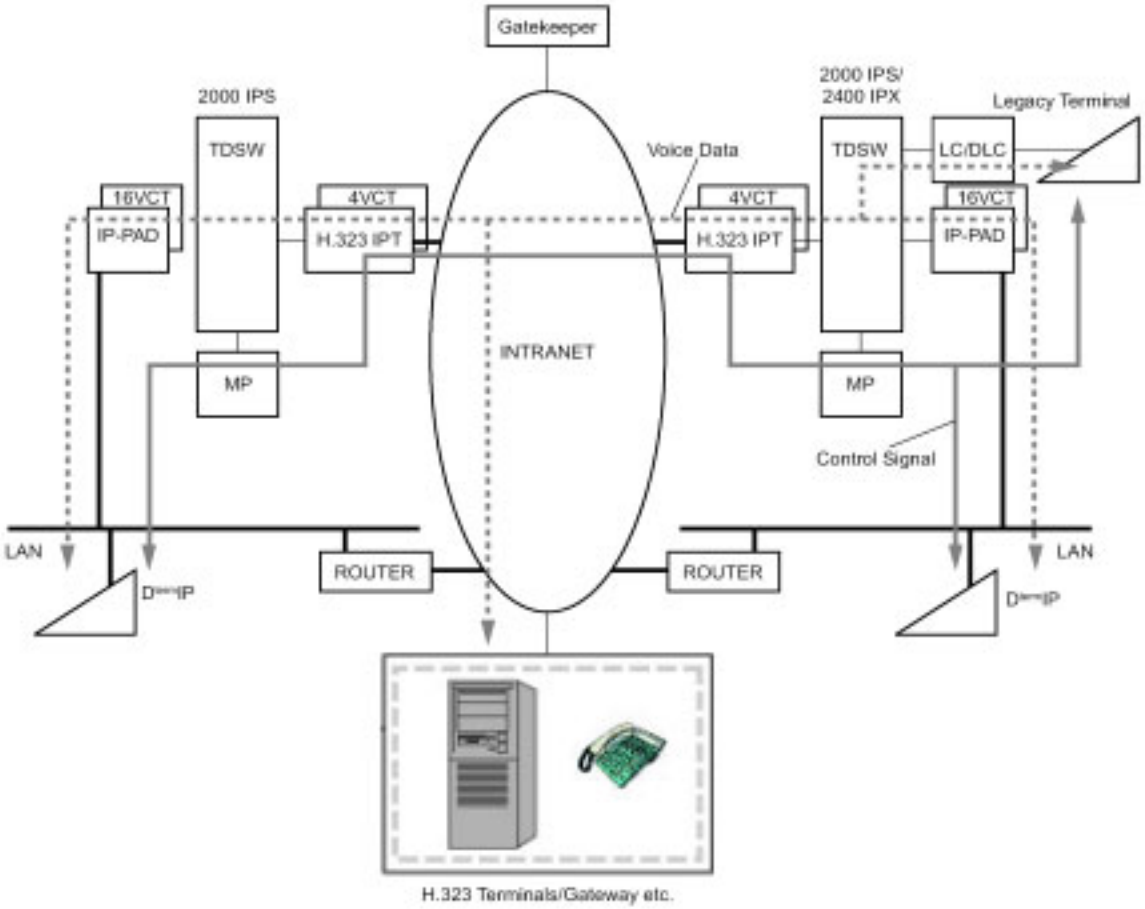
Required Equipment for IP CCIS via IP Trunk cards

Equipment Name	Functional Name	Function
SPN-IPTB-A PN-IPT	IPT	IP Trunk Card Accommodates the IP network and transmits/receives compressed voice or signals over IP network. 10 BASE-T/100 BASE-TX twisted pair cable is connected directly to this card.
SPN-4VCTI-A W/Cable	VCT	4-channel CODEC Card for IP Trunk (max of four per IPTB) Voice compression protocols: G723.1, G729A, G711, FAX (14.4 kbps),DTMF signals Used together with PN-IPTB card.

H.323 Connection

VoIP (Voice over IP) allows the system to transmit voice conversations over a corporate Intranet using ITU-T H.323 protocol. For DtermIP-to-DtermIP connection via the IP network with H.323 protocol, the IPT card and IP-PAD card are required to transmit and receive the control signal and voice data. For voice compression, the 16VCT card and 4VCT card are required. For Legacy terminal connection via the IP network with H.323 protocol, the IPT card is required.

System Outline of H.323 Connection



H.323 Features

The following service features are available when connecting with H.323 terminal (NEC product). For other vender products, confirmation test is required.

H.323 Features	
Call Forwarding-All Calls	Speed Calling-Station
Call Forwarding-Don't Answer	Bearer Service
Call Forwarding-Busy Line	Tandem Switching of TIE Trunks-2/4-Wire
Split Call Forwarding	The Line Connection with PAD Control
Call Transfer-All Calls	Outgoing Trunk Busy Announcement
Hotline	Brokerage Hot Line-Outside
Do not Disturb-Dterm	Least Cost Routing-3/6-Digit
Calling/Called Number Display	LCR-Time of Day Routing
Digital Display-Trunk	Attendant Manual Override
Speed Calling-System	Echo Canceller Control

Service Conditions for H.323

1. The data network used for H.323 must be a corporate Intranet.
2. Gatekeeper is required for H.323 network.
3. IP trunk cards must be connected with a corporate Intranet via a router.
4. CCIS feature key FD is required.
5. Maximum number of nodes in the network depends on the capability of the Gatekeeper.
6. Maximum one IP trunk card can be accommodated per PIM and maximum eight IP trunk cards per system.
7. Maximum voice channels per IP trunk card depends on the payload period as follows (payload period can be assigned in system programming):

Payload Period	Maximum Voice Channels per IP Trunk		
	G.729a	G.711	G.723.1
20 ms	6	5	–
30 ms	8	7	8
40 ms	12	10	–

Note: *Maximum 96 channels can be provided per system*

8. The same IP trunk card cannot be shared with CCIS network via IP and H.323.
9. Tone signals to an extension are provided by the system.
10. When a LAN cable is unplugged from the IP trunk card, the associated calls are disconnected.
11. When connected with the system by other manufacturer, a pre-connection test is required.
12. DTMF Relay is supported.

H.323 IP Trunk Specifications

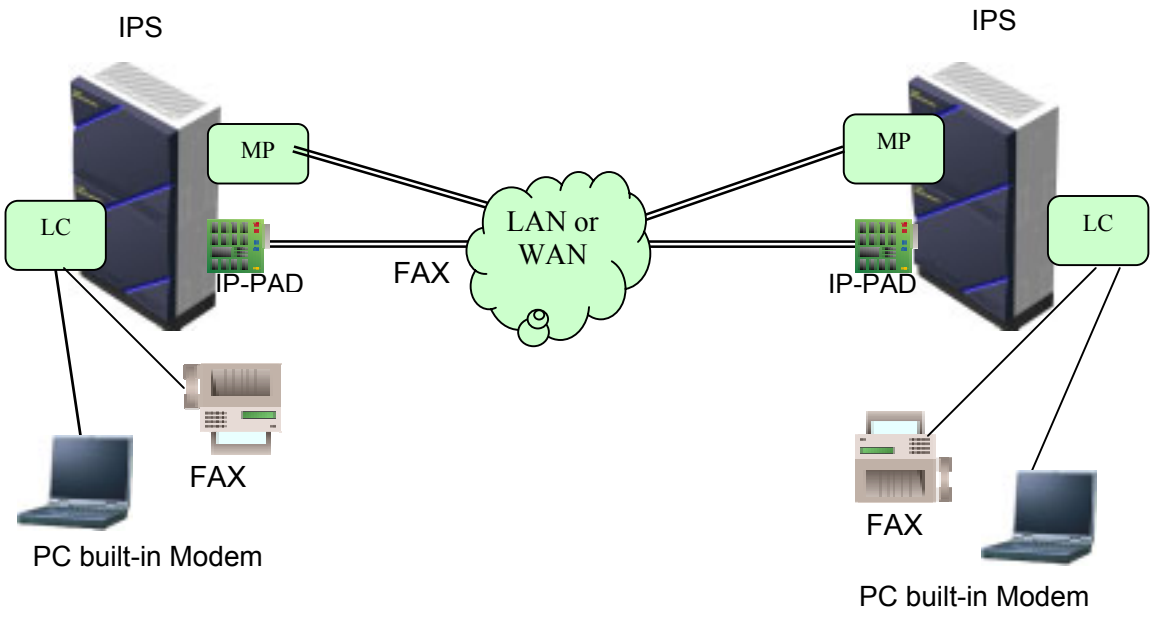
Description	Specifications
QoS signaling	IP Precedence/DiffServ (assigned on an IP trunk basis)
Speech encoding/decoding	G.711/G.729a/G.723.1
Payload size	G.711/G.729a : 20 ms, 30 ms, 40 ms G.723.1 : 30 ms (fixed)
Call model	Gatekeeper Routed/Direct
H.323 procedure	Normal Connect/Fast Connect
Supplementary service (defined in H.450)	Not available
Registration on Gatekeeper	Static registration
DHCP server	Not supported
FAX communications	Not available
PAD control	Controlled in a node basis
Echo canceller	Controlled in a node basis
Tandem connection	IPT to IPT/IPT to ACIS : available (but voice quality may deteriorate) IPT to CCIS : available IPT to ISDN : available IPT to analog C.O. : available IPT to MFC : not available
IPT connection mode	Point to Multipoint
Dynamic jitter buffer	10 to 600 ms (10 ms increment)

Required Equipment for H.323

Equipment Name	Functional Name	Function
SPN-IPTB-A PN-IPT	IPT	IP Trunk Card Accommodates the IP network and transmits/receives compressed voice or signals over IP network. 10 BASE-T/100 BASE-TX twisted pair cable is connected directly to this card.
SPN-4VCTI-A W/Cable	VCT	4-channel CODEC Card for IP Trunk (max of four per IPTB) Voice compression protocols: G723.1, G729A, G711, FAX (14.4 kbps),DTMF signals Used together with PN-IPTB card.

FAX and Modem over IP

This feature allows the system to transmit facsimile or modem communications over IP network, via Local Area Networks (LAN) and corporate Wide Area Network (WAN). Since PBX regards facsimile and modem equipment as one of ordinary telephones, IP Packet Assembler/Disassembler (IP-PAD) and Voice Compression Trunk (VCT) are required for facsimile uses over IP network same as legacy stations. The facsimile transmission procedure (T.30 or G.711/G.726 pass-through) is supported with IP-PAD/VCT. The following figure shows a typical configuration of facsimile or modem use on Peer-to-Peer CCIS network.



Example of FAX and Modem over IP connection

Service Conditions

1. IP-PAD and VCT are required for facsimile use on Peer-to-Peer CCIS network or Remote PIM over IP function.
2. PN-32IPLA (IP-PAD) card and PN-16VCTA (16VCT) card do not support FAX or Modem over IP. PN-32IPLA-A (IP-PAD) does not support Modem over IP.
3. IP-PAD card and 16VCT card support Fax over IP with the following protocol.

Cards	Protocol		
	G.711 Pass-Through	G.726 Pass-through	T.30
SPN-8IPLA IP PAD-C	X (Series 3300 software or later)	X (Series 3300 software or later)	–
SPN-8IPLA IP PAD-B	X (Series 3300 software or later)	X (Series 3300 software or later)	–
SPN-8IPLA IP PAD-A	X (Series 3300 software or later)	–	–
SPN-32IPLA IP PAD-E	X (Series 3400 software or later)	–	–
SPN-32IPLA IP PAD-E + SPN-16VCTAA IP PAD-B	X (Series 3300 software or later)	X (Series 3400 software or later)	X (Series 3200 software or later)
SPN-32IPLA IP PAD-E + SPN-16VCTAA IP PAD-A	–	–	X (Series 3200 software or later)
SPN-32IPLA IP PAD-C + SPN-16VCTAA IP PAD-A	–	–	X (Series 3200 software or later)

X: Available —: Not available

4. The analog Media Converter used by 2400 IPX cannot be accommodated in 2000 IPS. It is possible to connect facsimiles between 2400 IPX and 2000 IPS.
5. A problem may occur such as “expected transmission speed is not obtained” or “not connectable”, depending on a facsimile model even if it supports T.30.
6. If a Super G3 facsimile is used, the transmission speed will be equivalent to G3.
7. If a facsimile with Error Correction Mode (ECM) function is used, ECM does not work.
8. Connection speed for Modem over IP are:
   a. G.711 Pass-through 24kbps – 14kbps
   b. G726 Pass-though 14.4kbps – 9.6kbps

Note: *Due to many factors these speed are not guaranteed and an evaluation is suggested before actual use.*

Required Equipment for FAX or Modem over IP

Item	Description	Stock #	Qty	Remarks
1	PZ-M606-A	151492	1/Sys.	On board Ethernet Interface Card (ETHERNET TCP/IP)
2	SPN-32IPLAA IP PAD-E	153158	2/FP	IP Interface Card for IP PAD w/o FP: Max 2 w/ FP: Max 2/FP, Max 8/Sys.
3	SPN-16VCTAA IP PAD-B	153136	2/32IPLA	16ch Codec Card for IP PAD (Option) Max. 2/32IPLA Card
4	SPN-8IPLA IP PAD-C	153154	2/FP	IP Interface Card for IP PAD w/VCT
5	PZ-24IPLA	151254	1/8IPLA	Expansion Internet Protocol Line Circuit for IP PAD w/VCT

Connectable combinations between facsimile stations are shown below.

Connectable Combinations			Remarks
Source	**Via**	**Destination**	
IPS FAX station	Local	IPS FAX station	Station-to-Station connection
IPS FAX station	Peer-to-Peer CCIS	IPS FAX station	
IPS FAX station	Peer-to-Peer CCIS	IPS DM FAX station	
IPS FAX station	Peer-to-Peer Local	IPS DMR FAX station	Station-to-Station connection (Remote PIM over IP)
IPS FAX station	Peer-to-Peer CCIS	IPX FAX station	
IPS FAX station	Peer-to-Peer CCIS	IPX MC FAX station	

		Opposite Office (2000 IPS)						Opposite Office (2400/MC)
		8IPLA IP-PAD-B	**8IPLA IP-PAD-A**	**32IPLA IP-PAD-E**	**32IPLA IP-PAD-E + 16VCTAA IP-PAD-B**	**32IPLA IP-PAD-E + 16VCTAA IP-PAD-A**	**32IPLA IP-PAD-C + 16VCTAA IP-PAD-A**	
Own Office (2000 IPS)	**8IPLA IP-PAD-B**	G.711 G.726	G.711	G.711	G.711 G.726	N/A	N/A	G.711
	8IPLA IP-PAD-A	G.711	G.711	G.711	G.711	N/A	N/A	G.711
	32IPLA IP-PAD-E	G.711	G.711	G.711	G.711	N/A	N/A	G.711
	32IPLA IP-PAD-E + 16VCTAA IP-PAD-B	G.711 G.726	G.711	G.711	G.711 G.726 T.30	T.30	T.30	G.711 T.30
	32IPLA IP-PAD-E + 16VCTAA IP-PAD-A	N/A	N/A	N/A	T.30	T.30	T.30	T.30
	32IPLA IP-PAD-C + 16VCTAA IP-PAD-A	N/A	N/A	N/A	T.30	T.30	T.30	T.30

Required bandwidth for FAX Connection

• 32IPLA-A/16VCTA-A

Connection Conditions	Required Bandwidth (One-way)
T.30, G711, Payload=40ms	150 kbps (estimated)
T.30, G729a, Payload=40ms, Communication speed=14.4Kbps (No IP Header compression in Router)	23.6 Kbps (FAX Payload=78byte)
T.30, G729a, Payload=40ms, Communication speed=14.4Kbps (with IP Header compression in Router)	16.6 Kbps (FAX Payload=78byte)

• 8IPLA/24IPLA

Connection Conditions	Required Bandwidth (One-way)
G.711 pass-through, Payload=40 ms (No IP Header compression in Router)	72 kbps (FAX Payload=320 byte)
G.711 pass-through, Payload=40 ms (with IP Header compression in Router)	65 kbps (FAX Payload=320 byte)
G.726 pass-through, Payload=40 ms (No IP Header compression in Router)	40 kbps (FAX Payload=160 byte)
G.726 pass-through, Payload=40 ms (with IP Header compression in Router)	33 kbps (FAX Payload=160 byte)

Note: *The data in the above tables does not include MAC Header.*

Required bandwidth for Modem Connection

Connection Conditions	Payload length	IP Header compression in Router	Required Bandwidth (One-way)
G.711 pass-through	40 ms	No	72 kbps
	30 ms		74.67 kbps
	20 ms		80 kbps
	10 ms		96 kbps
G.726 pass-through	40 ms		40 kbps
	30 ms		42.67 kbps
	20 ms		48 kbps
	10 ms		64 kbps
G.711 pass-through	40 ms	Yes	65 kbps
	30 ms		65.33 kbps
	20 ms		66 kbps
	10 ms		68 kbps
G.726 pass-through	40 ms		33 kbps
	30 ms		33.33 kbps
	20 ms		34 kbps
	10 ms		36 kbps

Remote PIM over IP

Remote PIM over IP targets users who have 1-30 relatively small offices that accommodate 10-30 extensions at the Remote Site. When *UNIVERGE* NEAX IPS DMR or *UNIVERGE* NEAX 2000 IPS PIM are installed at remote site, and connected to an IPS or IPS DM at main site over IP network, the Main Site system controls and maintains the Remote PIM operation as one single system. If a communication failure occurs between the Main Site and Remote Site, the Remote Site automatically changes over to a survival mode and operates as a stand-alone system.

UNIVERGE NEAX IPS DMR: IPS Distributed Model Remote (with CP31-A/C)
UNIVERGE NEAX 2000 IPS: IPS PIMMK (with CP24-A/B/C)

The *UNIVERGE* NEAX IPS DMR is designed primarily for distributed IP networking but also supports traditional analog and digital trunks for connection to the Public Switched Telephone Network (PSTN). The *UNIVERGE* NEAX IPS DMR supports up to 128 peer-to-peer IP stations and 56 TDM ports in a single modular chassis. Up to two chassis can be stacked providing maximum capacity of 112 TDM ports while still supporting as many as 128 peer-to-peer IP stations.

Note: *The MP card at Remote Site has the same system data as the CPU at the Host Site; the Host Site automatically downloads system data to the Remote Site at the time of setup. In normal operation, Main Site automatically downloads a copy the system data to Remote Site through the network once a day.*

Because the CP31 as a DMR is designed as a Remote PIM CPU, the following options that are built-in on the CP24 are not available with the CP31:

- No built-in Modem
- No built-in DAT.
- Only one RS Port.
- No built-in DK (external/relay key).
- No MN Alarm Indication

Network Conditions and Payload

Item	Requirement	Remarks
Protocol	TCP/IP transparent	
Maximum Delay Time	120ms(one way)/240ms(return)Recommended 150ms(one way)/300ms(return) Maximum	Support the quality class A, B of IP Telephone

Bandwidth Requirement

Established Voice Calls		With G7.23.1 (5.3k/6.3k) Compression	With G729a (8k) Compression	Without Compression (G.711)
6	Control	4.1 Kbps	4.1 Kbps	4.1 Kbps
	Voice	31.8/37.8 Kbps	48 Kbps	432 Kbps
8	Control	4.3 Kbps	4.3 Kbps	4.3 Kbps
	Voice	42.4/50.4 Kbps	64 Kbps	576 Kbps
12	Control	4.3 Kbps	4.3 Kbps	4.3 Kbps
	Voice	63.6/75.6 Kbps	96 Kbps	864 Kbps
16	Control	4.5 Kbps	4.5 Kbps	4.5 Kbps
	Voice	84.8/100.8 Kbps	128 Kbps	1152 Kbps
24	Control	4.5 Kbps	4.5 Kbps	4.5 Kbps
	Voice	127.2/151.2 Kbps	192 Kbps	1728 Kbps
32	Control	4.9 Kbps	4.9 Kbps	4.9 Kbps
	Voice	169.6/201.6 Kbps	256 Kbps	2304 Kbps
48	Control	4.9 Kbps	4.9 Kbps	4.9 Kbps
	Voice	254.4/302.4 Kbps	384 Kbps	3456 Kbps
64	Control	5.8 Kbps	5.8 Kbps	5.8 Kbps
	Voice	339.2/403.2 Kbps	512 Kbps	4608 Kbps
72	Control	5.8 Kbps	5.8 Kbps	5.8Kbps
	Voice	381.6/453.6 Kbps	576 Kbps	5184 Kbps
96	Control	6.7 Kbps	6.7 Kbps	6.7 Kbps
	Voice	508.8/604.8 Kbps	768 Kbps	6912 Kbps

Note: *This information is an estimation based on an established call. Slightly Higher Control values will occur at time of call origination and termination.*

Base values

- Originating from a station: 9.6 Kbps/Call (estimated)
- Terminating to a station: 5.76 Kbps /Call (estimated)
- Originating to C.O: 11.5 Kbps/Call (estimated)
- Terminating from C.O: 5.76 Kbps/Call (estimated)
- Keep Alive to Remote Site: 0.032Kbps (estimated)
- Other control packets for Remote Site: 4Kbps (estimated)
- G.723.1 voice: 5.3Kbps (one-way)
- G.729a voice: 8Kbps (one-way)
- G.711 voice: 64Kbps (one-way)

The above base values are primarily used for call setup with the exception of keep alive; 0.032Kbps with no voice traffic. Connections between IP PAD are half duplex, established call utilization is G.711 voice: 64Kbps, G.723.1 voice: 5.3/6.3Kbps, or G729a voice: 8Kbps. Peer-to-Peer IP station calls are full duplex, compression can be specified by location numbers in system data. Peer-to Peer IP station calls even though full duplex will utilize one-way for Bi-directional networks such as T1. Peer-to Peer IP station calls over Asymmetrical networks such as ADSL may realize higher bandwidth utilization, compression can be specified by location numbers in system data.

Advantages

The system regards the terminals accommodated in both Host Site and Remote Site as the extensions in the same office. Feature transparency is superior to CCIS.

The Digital Remote PIM cannot accommodate some AP cards; Remote PIM over IP can accommodate AP cards such as ISDN PRI and T1.

This feature can reduce the bandwidth used on the WAN that is connected to CO lines at Remote Site, rather than DtermIP at remote locations.

Since all Remote PIM over IP sites are treated as extensions in the same office, software and applications only have to be implemented in the host site. This provides centralized use of application for example distributing ACD agents in the DMR locations. CCIS requires each location to have separate software and applications.

CCIS over IP can be combined with Remote PIM over IP to accommodate larger network configurations. Up to 255 host sites can be connected via CCIS, each host site can have up to 15 Remote PIM over IP locations.

User Mobility/User Continuance allows a user to relocate from one site to another then login using the same station number and password. User Mobility/User Continuance is only available to the 2000 IPS in a Remote PIM network using a DtermIP terminal.

Service Conditions

1. It is a requirement that the same software version be loaded in the Main Site and All Remote Sites.

2. Host site can be an IPS, IPS DM or IPS Retro system. Main plus Remote PIM over IP is available in any combination of the following CPUs.
   Main Site: CP24-A/B/C, CP27-A, CP26-A, CP28-A
   Remote Site: CP31-A/B/C, CP24-A/B/C, CP27-A, CP26-A, CP28-A
   Note: *The CP27/CP28 can not be used as Backup CPU system in a Remote PIM.*

3. Software and Key FD for the whole system must be loaded at the Main Site. No software or key's can be loaded into the Remote Site.

4. All system data changes for the whole system must be performed in the Main Site. No system data changes can be done in the Remote Site.

5. A TCP/IP network is required between the Main Site and Remote Site. A closed and bandwidth guaranteed network is preferable, such as IP-VPN (layer 3 VPN) or wide area Ethernet service (layer 2 VPN). Permissible Delay with in these types of networks is 100ms one way (120ms Max) or 200ms both way (240ms Max).

6. The CPU card at Remote Site has the same system data as the CPU at Main Site; the Host Site automatically downloads its system data to the Remote Site at the time of setup. In normal operation, Host Site automatically copies the system data to Remote Site through the network once a day.

7. Remote Site automatically operates by itself (survival mode) when Keep Alive signal (sent every 30 sec) between the Host Site and Remote Sits is interrupted. When Keep Alive is interrupted the Remote Site is reset to change the operation from normal mode to survival mode.

8. Remote Site in survival mode checks at 30 seconds intervals if the communications to Main Site are possible. When Keep Alive is detected, the Remote Site automatically is reset to change the operation from survival mode to normal mode.

9. When unstable conditions occur in the network, the Remote Site can be manually set to survivable mode (override automatic) until stability in the network is established. This prevents the Remote Site from resetting normal mode to survivable mode etc.

10. User Mobility/User Continuance is only available to IP enabled Dterm, DtermIP and Soft Phone Dterm SP30 terminals using Protected Login Mode. The SP30 is not supported when used in collaboration mode with a Dterm or wireless PS.

11. For User Mobility/User Continuance to operate at the visitor site (Main or Remote site), a virtual station port for each visitor station is required. Each visitor port assigned reduces the total number of virtual FP ports available to that site (Main or Remote Site), which reduces the total number DtermIP terminals.

12. While logged in as a visitor the location number, tenant number and service features depend on the setting of the site that the terminal is logged into. At the time if login at the visitor site stack dial and last number redial are erased.

Required Hardware and Software

Host Site

Equipment Name	Remarks
PZ-M606-A	On board Ethernet Interface card
SPN-8IPLA IP PAD	8 Port PAD with built-in compression
PZ-24IPLA	24 Port PAD Expansion, mounts on SPN-8IPLA
SPN-32IPLAA IP PAD	Provides Packet assembly/disassembly to accommodate Legacy Line/Trunk interface.
SPN-16VCTAA IP PAD-B	for compression or T.30 FAX (Optional)
R-PIM 1 Site License	1 required for each Remote site

Note: *Registration of Host CPU and software required*

Remote Site

Equipment Name	Remarks
PZ-M606-A	On board Ethernet Interface card
SPN-8IPLA IP PAD	8 Port PAD with built-in compression
PZ-24IPLA	24 Port PAD Expansion, mounts on SPN-8IPLA
SPN-32IPLAA IP PAD	Provides Packet assembly/disassembly to accommodate Legacy Line/Trunk interface.
SPN-16VCTAA IP PAD-B	for compression or T.30 FAX (Optional)

Note: *Registration of software is "not" required. Core software of remote site should be the same version as main site*

List of Service Features Operation

X: Available —: Not Available

Service Features	Main Site	Remote site	
		Normal Mode	Survival Mode
Multiline Terminal, SLT operation	X	X	X
IP Enabled Dterm operation	X	X	X
COT, LDT, ODT	X	X	X
Attendant Console	X	—	—
Add-On Module	X	—	—
DSS Console	X	X	X
ISDN Terminal	X	—	—
IPT (CCIS, H.323)	X	—	—
Built-in IPT	X	—	—
OAI client	X	X	—
PS, ZT	X	—	—
Trunk IC/OG call (CO, Tie line)	X	X	X
Caller ID Class	X	—	—
Caller ID Display (MFC, T1-ANI)	X	—	—
Caller ID Display (ISDN)	X	X	X
Caller ID-Station	X	—	—
Enhanced 911	X	—	—
CCIS (Digital, Analog)	X	X	X
SMDR	X	—	—
MAT (RS232C)	X	X	X
MAT (LAN)	X	X	X
MAT (Modem)	X	—	—
System Data change by MAT	X	—	—
SNMP	X	—	—
VLAN	X	X	X
Fault Message registration	X	—	X
Announcement Service	X	—	—
Automated Attendant	X	—	—
Background Music	X	—	—
Conference (Three/Four Party)	X	X	X
Conference (Six/Ten Party)	X	—	—
Conference (32 Party)	X	X(**Note 2**)	—
Internal Zone Paging	X	—	—
User Mobility/User Continuance **Note1**	X	X	X

Note1: *User will be required to login in again once remote system starts in survival mode.*
Note2: *Available using CFTC card in the Main Site.*

Planning and Installation

Vendor Support for Open System Standards

The vendor should be committed to supporting open system industry standards, such as H.323, 802.1p and 802.1q, MGCP, TAPI, JTAPI, *etc.*

Voice Messaging:

A scalable and cost-effective voice messaging solution that supports industry standards such as AMIS-A, VPIM, LDAP and IMAP.

Bandwidth requirements

One of the most important factors in the success of your IP network is to determine the necessary network bandwidth needed for your application. Determining the appropriate bandwidth is truly a function of two factors:

- Number of Users (Phones)
- Voice Compression for each user

Number of Users: The number of users is the number of simultaneous users (digital phones) that will be required.

Voice Compression: NEC's IP solutions deploy voice compression in order to extend multiple users across fewer data channels. The following compression algorithms are supported: G.711, G.723.1 and G.729A. Depending on the voice compression algorithm selected and the number of voice frames encapsulated in one data packet.

Select the Proper Voice Compression: The best voice quality is achieved by using the non-compression voice algorithm G.711. The maximum quality comes at the expense of the highest utilized bandwidth. The largest vice compression is achieved by using G.729A. If you are using this algorithm, you will save on bandwidth and still achieve voice quality that is regarded as near toll. If absolute conversation quality is your focus and bandwidth is no object, you probably want to select G.711. If bandwidth is a priority you will need to employ G.729A.

Network Consideration

There are several network considerations that must be met before deployment of IP Telephony. These items are:

- The network must support the Internet Protocol and the local interface must meet 802.3 Ethernet standards.
- It is recommended that the managed network be based on Switched, Fast Ethernet environment for IP telephony.
- Packet Delay or Latency, one-way should be less than 100 milliseconds optimal, with a maximum of 250 milliseconds one-way.
- Jitter, or out of sequence packets, should be between 10 -150 milliseconds optimal with a maximum of 300 milliseconds while packet loss of 1% or less is recommended with the maximum of 3% from end to end.

Network Bandwidth

Network bandwidth requirements are based on the total number of IP trunks or IP stations installed in the PBX. The multiple algorithms available the bandwidth needed for a voice call out to an IP network can range from 10.7 Kbps to 96 Kbps.

The most common algorithm used for NEAX IP trunks is G.729A, which allows the voice to be compressed to 8KBps. Once the layer 3 (IP) overhead is added to the voice payload, the approximate bandwidth is 16KBps for a single voice stream out of the PBX to an IP network. This bandwidth calculation does not include layer 2 overhead and will vary depending on the type of transport (Frame Relay, ATM, Ethernet etc.). Depending on what type of algorithm is in place and how many devices are deployed dictates the amount of bandwidth needed.

The bandwidth for call setup for the CCIS signaling channel over IP is 7.2 Kbps per call and the FCCS signaling channel is 83.2Kbps per call. Call teardown requires the same amount of bandwidth. This number varies and 7.2 Kbps and 83.2 Kbps are averaged values from the start sequence until setup or teardown is accomplished.

This means to make an FCCS or CCIS call from PBX A to PBX B, PBX A will transmit 83.2 Kbps (FCCS) or 7.2 Kbps (CCIS) as the F/C channel until the call is established. Once the call is established, PBX A will stop sending the 83.2 Kbps/7.2 Kbps setup information and will begin sending the appropriate voice payload per the encoding/decoding algorithm selected (G.711, G.729A, G.723.1).

To make multiple calls, the system will send 7.2 Kbps/83.2 Kbps for each call until all the calls are established. During tear down 7.2 Kbps/83.2 Kbps will be transmitted for each call again.

If an IP station has been idle for up to 4 seconds, a 7.6Kbps “Keep Alive” packet will be sent to those respective devices. A “Keep Alive” packet is not generated to an idle station if it is configured across a WAN. This same process occurs on IP trunk routes. A “Keep Alive” packet is sent to the trunk route for each system or Point Code in the network. If traffic is present on either IP stations or trunk routes, no “Keep Alive” packet is generated.

Bandwidth utilization

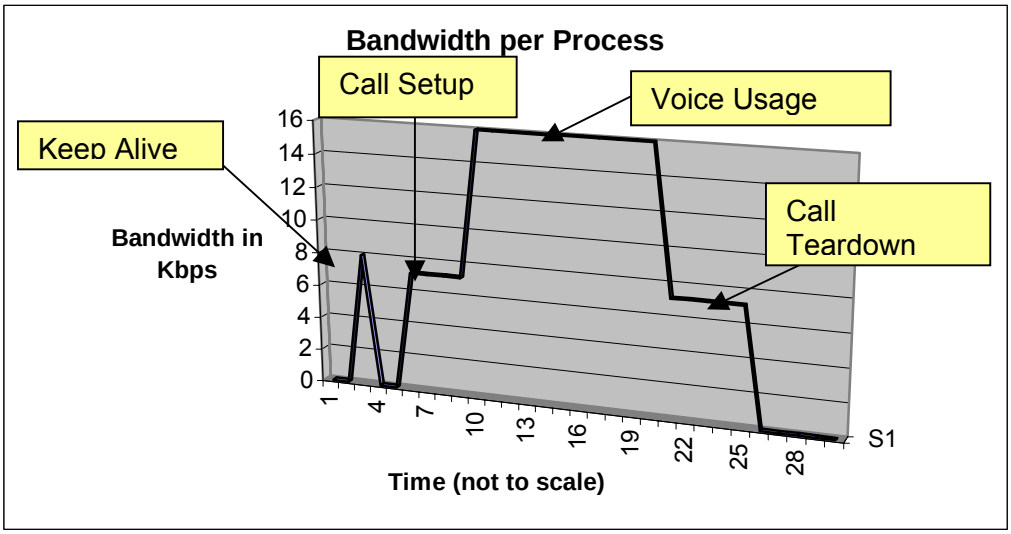
The following chart shows the amount of bandwidth for IP overhead per the fill times set in the PBX system. The higher the fill time used, better performance from the PBX and IP network can be realized due to the smaller number, yet larger (in size) packets generated.

Codec	Filler Time			
	10 msec	20 msec	30 msec	40 msec
G.711 (64 Kbps)	32 Kbps	16 Kbps	10.67 Kbps	8 Kbps
G.729a (8 Kbps)	32 Kbps	16 Kbps	10.67 Kbps	8 Kbps
G.723.1 (6.3 Kbps)	N/A	N/A	10.67 Kbps	N/A
G.723.1 (5.3 Kbps)	N/A	N/A	10.67 Kbps	N/A

This chart shows the total amount of bandwidth in layer 3 per voice call during the transmit stream from an IP trunk or IP station. The value shown is based on the fill time from the table above and the bandwidth required by the encoding/decoding algorithm that is used (codec).

Codec	Total Bandwidth per transmit stream			
	10 msec	20 msec	30 msec	40 msec
G.711 (64 Kbps)	96 Kbps	80 Kbps	74.67 Kbps	72 Kbps
G.729a (8 Kbps)	40 Kbps	24 Kbps	18.67 Kbps	16 Kbps
G.723.1 (6.3 Kbps)	N/A	N/A	16.97 Kbps	N/A
G.723.1 (5.3 Kbps)	N/A	N/A	15.97 Kbps	N/A

This chart shows bandwidth usage (G.729A) based on an idle state followed by a keep alive message with a call setup and voice usage next. Time is not to scale.



Quality of Service (QoS)

In order to obtain quality of service (QoS) for the voice traffic, routers or layer 3 switches in the network must support a queuing mechanism. The queuing mechanism allows for the prioritization of the voice packets to pass through the router before the routine data traffic. The NEAX PBX system allows for customizing the priority of the voice traffic out of the PBX. This prioritization occurs with the IP precedence bits inside the IP header. With a higher level of prioritization, the voice traffic will not accumulate as much delay as traffic with lower levels of priority. With Diff-Serve (Differentiated Services), prioritization can be made if the routing devices support it. NEAX PBX IP trunks support Diff-Serve.

QoS should be pushed as close to the network edge as possible. At the edge, this is considered CoS (Class of Service). A layer 2 switched environment is recommended to obtain full throughput to a routing device for NEAX IP telephony and for CoS management.

Specific ports on the data switch may be prioritized for QoS or CoS. If this is available, the port that is connected to a NEAX IP telephony device should be prioritized in this manner. Additionally in the network, the port that has been prioritized, in the above manner, must also be set as a trusted value in the network. When CoS is used on the edge, map it to a higher level (layer 3) QoS value. This mapping allows for a consistent prioritization throughout the entire network. By using the prioritization methods described for the PBX and the network optimal data transfer in the core and edge devices throughout the network may be accomplished.

Device Registration Server (DRS) and DHCP

The Device Registration Server (DRS) is built into the MP card (System-based DRS). The System-based DRS can provide log-in/log-out function and registration authorization function of IP Enabled Dterm terminals. Up to 956 IP Enabled Dterm terminals can be managed by the System-based DRS. Once the Device Registration Server (DRS) is located, depending on the configuration chosen, the IP Enabled Dterm will prompt for a login and password. Network-based DRS (external DRS) 3.0 is not supported.

DHCP (Dynamic Host Configuration Protocol) can be used for ease of administration for IP Enabled Dterm terminals in the network. The IP adapter can be set to use a DHCP server to supply its IP address and the address of the DRS. Network administrators can set up the option in the DHCP server to give the IP Enabled Dterm terminal the information needed to find the DRS.

TCP/UDP/RTP PORT NUMBER CONDITIONS

Tables below show the default TCP/UDP/RTP port number and range of the available TCP/UDP/RTP port numbers used in 2000 IPS

For Dterm IP

Port Type	Destination	Default Port No.	Remarks
Port of Login to DRS (UDP)	2000 IPS (MP Built-in DRS of Main Site)	3456	1024-65535 Assign by CM0B Y=10>60
	2000 IPS (MP Built-in DRS of Remote Site)	3456	Fixed
	Dterm IP	3455	1024-65535 Assign by Dterm IP set up (User Menu)
Port for PROTIMS (UDP)	2000 IPS	50000 + 64 x Virtual FP No. + Dterm IP Port No.	Base Port: 1024-65534 Assign by CM0B Y=10>00-59 (Virtual FP No.) (Default: 50000 + 64 x Virtual FP No.)
	Dterm IP	3458	1024-65534 Assign by Dterm IP set up (User Menu)

For Voice Control/Voice Packet

Port Type	Destination	Default Port No.	Remarks
Port for Voice Control (UDP)	Dterm IP	4000	1024-65535 Assign by Dterm IP set up (User Menu)
	IP PAD	50000	1024-65534 Assign by CM0A Y=10-17/ 30-37/100-115>90
Port for H.245 Voice Control (TCP)	Dterm IP	0	1024-65534 Assign by Round Robin method (Use From the lowest in blank port)
	IP PAD	6000-7024	Base Port: 1024-65534 Assign by CM0A Y=10-17/ 30-37/100-115>92 (Set every 1025 port)

For Voice Control/Voice Packet

Port Type	Destination	Default Port No.	Remarks
Port for Voice Packet (RTP)	Dterm IP	3462	1024-65534 Assign by Round Robin method (Use From the lowest in blank port)
	IP PAD	9998-10317	Base Port: 1024-65534 Assign by CM0A Y=10-17/ 30-37/100-115>93
	IPT (CCIS)	56000-56062	Base Port: Fixed to 56000
	IPT (H.323)	56000-56047	Base Port: 20000-65000 Assign by CMBA Y=34 (Default: 56000)

For CCIS Virtual IPT

Port Type	Destination	Default Port No.	Remarks
Port for Server PBX (TCP)	Server PBX	57000	1024-65534 Assign by CM0B Y=20>00
Port for Client PBX (UDP)	Client PBX	58000-59023	Base Port: 1024-65534 Assign by CM0B Y=20>01 Select by Round Robin method (Use from lowest in blank port)

For H.323 IP Trunk

Port Type	Destination	Default Port No.	Remarks
H.255.0 RAS Port (UDP)	IPT	1719	Fixed to 1719
H.255.0 Control Port (UDP)	IPT	1720	Fixed to 1720

For Voice Control/Voice Packet

Port Type	Destination	Default Port No.	Remarks
Port for Remote PIM Control (UDP)	2000 IPS (Main Site)	3300	1024-65534 Assign by CM0B Y=10>70 DHM Self Port (Communication port between the Main Site and Remote Site), Common Port
	2000 IPS (Main Site)	3300 + Remote Site No.	Port per Remote Site (MP-MP) Port per Remote Site: Setting data of CM0B Y=10>70 + Site No.
	2000 IPS (Main Site)	50000 + FP/AP No. of Remote Site x 64	1024-65534 Assign by CM0B Y=10>00-59 (Virtual FP No. of Remote Site) Port per Remote Site (MP-FP)

System Conditions/Limitations (Peer-to-Peer IP)

Limitations

Add-on-Module:
Dterm IP does not support Add-on Module feature.

Analog Port Adapters:
When the IPW-2U unit is attached with a Dterm Series E terminal, other optional adapters (except for ADA-U Unit) cannot be attached with the Dterm Series E terminal.

Back Ground Music (BGM):
Service features requiring continuous voice transmission such as Background Music feature cannot be used because the traffic may reduce overall performance of the Local Area Network (LAN).

Internal Zone Paging:
There is a maximum of three DtermIPs per Internal Zone Page Group. Internal All Zone Paging will not access IP Dterm.

Set Relocation:
Set Relocation is not available between Dterm IP terminals. This feature is for legacy terminals only.

Note: *Legacy stations/trunks means the ones connected to the Time Division Switch (TDSW). Examples of legacy station/trunk cards are LC, DLC, COT, PRT, CCT, DTI, 8RST (DTMF receivers), CFTB (conference trunk), etc.*

Conditions

DSS/BLF:
A DSS/BLF Console can be associated with the Dterm IP and can be used in a Remote PIM, but the DSS/BLF console must be connected to DLC card.

IP-PAD:
The IP-PADs are required for the following connections/statuses:

- Connections between Dterm IP terminals and legacy stations/trunks
- Connections between Dterm IP terminals and IP trunks (H.323)
- Connections between legacy stations/trunks and IP trunks (H.323)
- Connections for CCIS networking via IP from/to legacy stations/trunks
- While Dterm IP terminals are on hold (Consultation Hold, Call Transfer, Music-on-Hold, etc.)
- When any override service is activated (Executive Override, etc.)
- Three/four-party conference including Dterm IP terminals

Chapter 14 Common Channel Inter-Office Signaling (CCIS)

The *UNIVERGE* NEAX 2000 IPS offers a very powerful proprietary networking feature called Common Channel Inter-Office Signaling (CCIS). This functional and extremely flexible "intelligent network" capability allows two or more *UNIVERGE* telecommunications systems to be networked together to provide feature transparency and to centralize many important telecommunications functions such as Attendant Consoles, Call Accounting / Billing Systems, Voice Processing Systems, Management systems, and Trunk Facilities, to name a few.

A variety of NEC's *UNIVERGE* family of telecommunications systems can be networked together using CCIS, creating a proprietary corporate network. This network can serve multiple company facilities in the same building or town, in different towns, in different states, and even in different countries. Centralized services provide a great cost reduction in the initial system hardware costs, ongoing maintenance costs, trunk and tie line costs and overall operations costs. Feature transparency increases user productivity by providing a common numbering plan for simplified on-net calling and a common feature package. Feature transparency allows most features to be accessed using similar procedures throughout the local, national, or global corporate network.

Alternate Routing capabilities provide increased network reliability, additional cost savings, and increased user productivity. Based on the industry standard CCITT Signaling System #7, which is the frame work for ISDN, CCIS offers 64k "Clear" channel voice/data transmissions. The CCIS "Intelligent Network" can be configured to provide redundant signaling channels and multiple alternate routing schemes for maximum network reliability. CCIS is so flexible it can be used with standard digital T1 spans, analog tie lines, or satellite transmission systems.

The CCIS architecture is similar to ISDN. Both use 23B + D type trunks. Both networking types conform to and comply with the Open Systems Interconnection (OSI) 7 Layer Model.

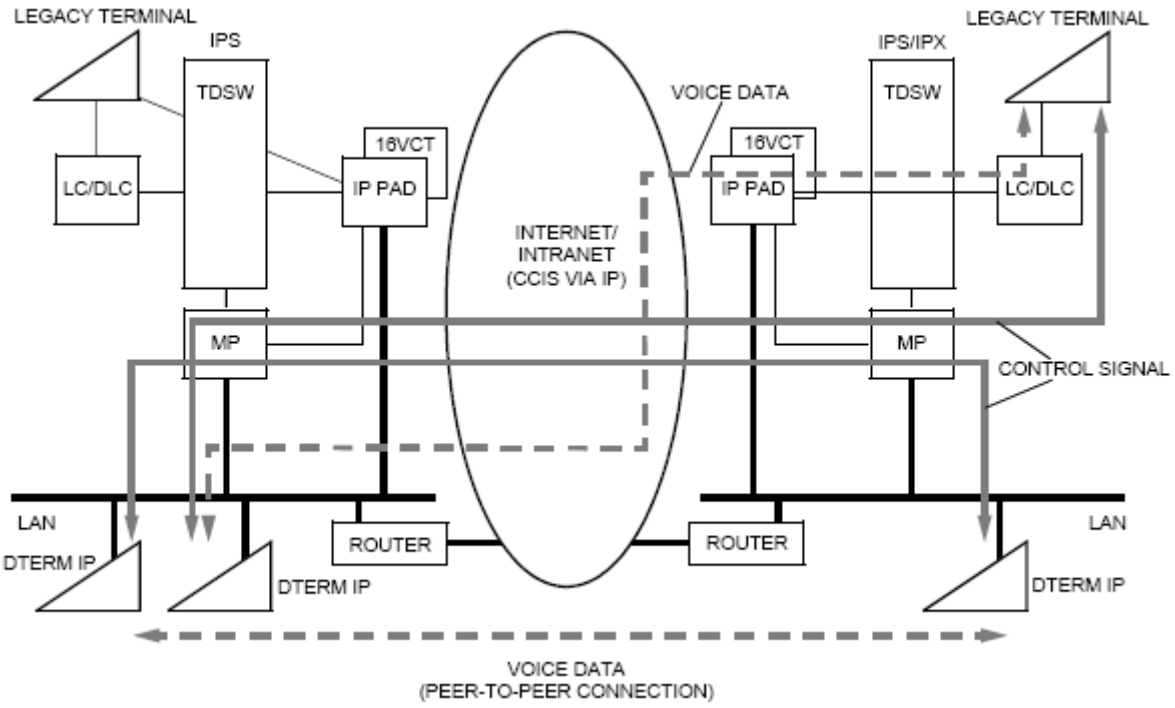
The *UNIVERGE* NEAX 2000 IPS can be interfaced to another *UNIVERGE* NEAX PBX by No. 7 CCIS Signaling (Analog/Digital), CCIS Networking via IP (non Peer-to-Peer connection), or via IP (Peer-to-Peer connection). For adding No. 7 CCIS to the system, it is necessary to install the 24 channel DTI (Digital Trunk Interface) for a digital network or LDT (Loop Dial Trunk)/ODT (2 wire E&M or 4 wire E&M Trunk) for an analog network via a MODEM, and a PLO (Phase Locked Oscillator) for network synchronization. Also, it is necessary to install a CCH (Common Channel Handler). The CCH receives/transmits common signaling data to/from the distant office. In each local office, the PBX can provide Centralized Billing function in addition to a variety of inter-office service features. For addition of the Centralized Billing function, an AP (Application Processor) is required when the *UNIVERGE* NEAX 2000 IPS is the center office and Built-in SMDR when the *UNIVERGE* NEAX 2000 IPS is used as a tandem or local office.

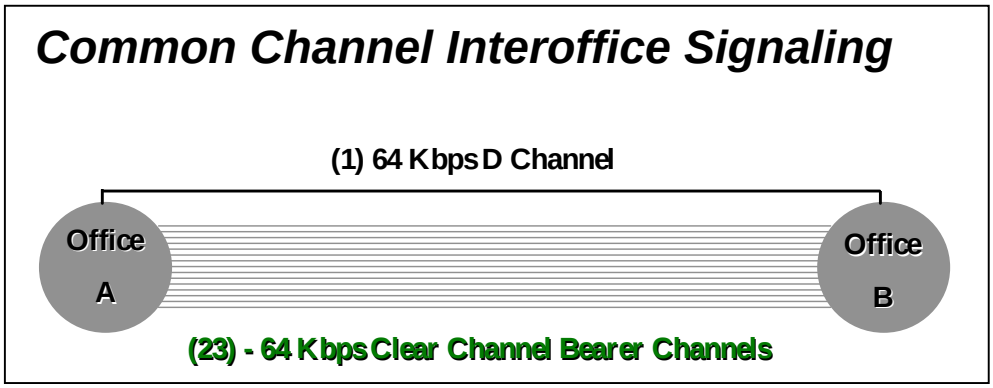
Note: *Centralized Day/Night Mode Change and Centralized Fault Message require CCIS No.7 networking with the IPX.*

For DtermIP to DtermIP connection via CCIS (Peer to Peer connection), the voice data is transmitted and received directly between DtermIPs via Intranet (CCIS via IP). For Dterm IP to Legacy terminal connection via CCIS, the IP-PAD card is required to transmit and receive the voice data. This card is used to control and convert the voice to data. The control signals are managed by the MP card in either of the connection above.

The illustration below shows a system outline of CCIS Connection.

System Outline of a CCIS Connection





Signaling

CCIS uses a common channel to provide the signaling information between two systems (nodes). The information transmitted in the signaling channel between two systems includes addressing information, supervisory information, and centralized network information. This signaling information is to be transmitted between all network nodes for a fully integrated network.

Addressing Information

This information includes dialed digits (called number) and calling party identification. CCIS provides the ability to provide calling number identification or the name of the calling party to any display station across the network. Users can screen calls by looking at the display prior to answering the call.

Supervisory Information

This information includes station status, call set up and termination information.

Network Information

This information includes centralized billing information, system fault data messages, message waiting indication control information, voice processing integration information, and data related to other network feature control.

CCIS and ISDN

Since the first CCIS system was installed in 1985, CCIS has proven to be the most reliable and versatile proprietary network system available. The delays in achieving nation wide ISDN compatibility, and the fact that a total global ISDN network is not yet available, exemplify the true values of an intelligent proprietary CCIS network. Additionally, it is easy to understand that a global network that only needs to be compatible with one family of telecommunications products can easily offer much more feature transparency and centralization than a global network that is intended to be compatible with all telecommunications systems. For this reason alone, the benefits of a CCIS network are much greater than those of an ISDN network. And with CCIS, it is possible to create a mixed network of CCIS and ISDN.

Digital and Analog CCIS

The PBX can provide No. 7 CCIS via either a digital network or an analog network. Regardless of the relevant network being a digital network or an analog network, CCH (Common Channel Handler) to control the common signaling channel is required.

Digital CCIS

Digital CCIS uses standard 24 channel digital T1 facilities to provide 23 B (voice/data) channels and 1 D (signaling) channel for the first CCIS dedicated T1 span. Additional T1 spans can be added and all 24 channels of each additional T1 span could be used for voice and/or data transmission. In this case, the D channel of the first T1 span is using non-facilitated associated signaling for the additional T1 spans. A second D channel of the first T1 span can be configured to provide a redundant signaling link between nodes to increase network reliability. To provide even more reliability, one channel of a second T1 span can be programmed to serve as the redundant signaling link. In this configuration, network reliability is at a maximum because if the first T1 span fails, the 23 B channels and the redundant D channel of the second T1 span would keep the network operational.

The D channel can be programmed to transmit the signaling information at rates of 48K, 56K, or 64K bits per second. At these high transmission rates, one D channel can easily handle the signaling requirements for multiple T1 spans (non-facilitated associated signaling).

When using a Satellite system as the transmission medium, the NEAX Family of systems are usually configured for digital CCIS, however, it should be noted that many Satellite systems can be configured to accept multiple analog tie lines which would be compatible with analog CCIS.

Analog CCIS

For applications requiring analog tie lines, CCIS requires the use of modems at each end of the dedicated signaling tie line. These modems can operate as high as 56K baud rate. The voice and data lines (B channels) also use standard analog tie lines. If the customer requires 7 voice / data channels between nodes, 8 tie lines are ordered providing one extra tie line to be dedicated as the signaling channel. This may appear to require more tie lines then a non-CCIS network but actually less tie lines are required when CCIS is used thanks to the intelligent Look Ahead Routing feature.

IP CCIS

Peer-to-Peer Connections

Peer-to-peer connections over CCIS Networking via IP are available when the distant systems are 2000 IPS or 2400 IPX supporting peer-to-peer connections. The built-in IP trunks (CCH) on the MP card are assigned with the Virtual Application Processors (AP), and can support up to 127 trunks per system. One Virtual AP can support up to 64 trunks, and up to two Virtual APs can be assigned per system.

The 8 IPLA IP PAD is required for connections between legacy stations/trunks, IP terminals and legacy stations/trunks over the CCIS networking via IP. The IP-PAD converts voice packet data to PCM signals, and one IP-PAD can provide 8 PCM channels or one 8 IPLA plus one 24 IPLA can provide 32 PCM channels. The number of IP-PADs depends on the traffic volume of those connections, and up to eight 8 IPLA IP PADs with 24 IPLA daughter boards can be accommodated per system, thus providing 256 PCM channels in total. Up to two IP-PADs can be controlled by the MP without FP cards, and additional IP-PADs can be controlled by the FP cards (two IP-PADs per FP card).

Note: *Legacy stations/trunks means the ones connected to the Time Division Switch (TDSW). Examples of legacy station/trunk cards are LC, DLC, COT, PRT, CCT, DTI, etc*

Voice Compression

Voice compression is available for CCIS networking via IP to/from legacy stations/trunks.

The following type of voice compression is available:

G.711 (64Kbps)
G.729a (8Kbps)
G.723.1 (5.3Kbps/6.3Kbps)

Voice compression can be assigned on a call basis or terminal basis, by system programming.

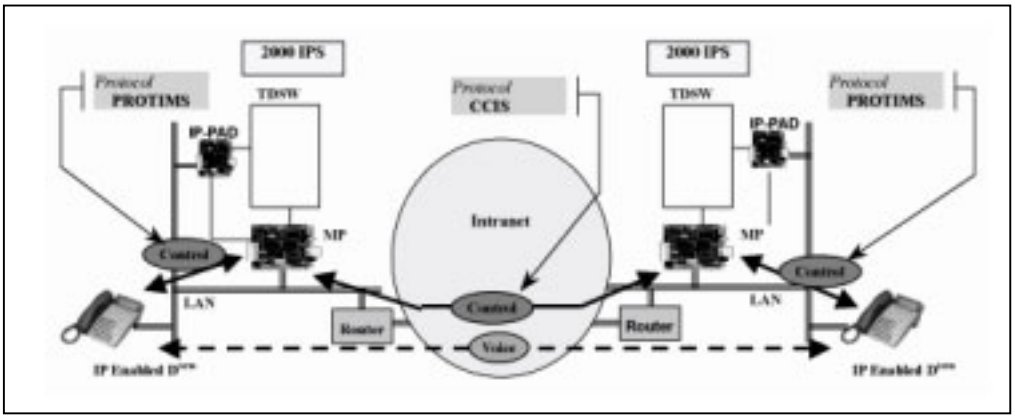
The 8 IPLA IP-PAD and the 24 IPLA daughter board both have built-in VCT capability to provide voice compression which is required for G.729a and G.723.1.

Service Conditions

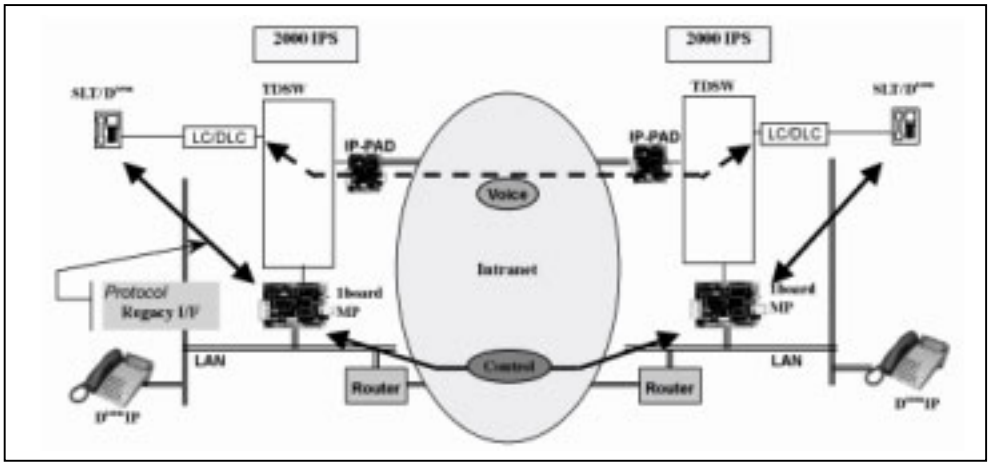
- Intranet must be used for a CCIS network via IP. (Internet is not supported.)
- Maximum number of nodes in the CCIS network via IP is 255.
- All CCIS features are available in the CCIS networking via IP.
- When T1 timeout of IAI is occurred, an outgoing CCIS call can be routed over alternate trunk route. This alternate routing can be allowed or denied on a trunk route basis by system programming.

Note: *The alternate routing is not available for tandem calls.*

CCIS Networking via IP between IP Terminals

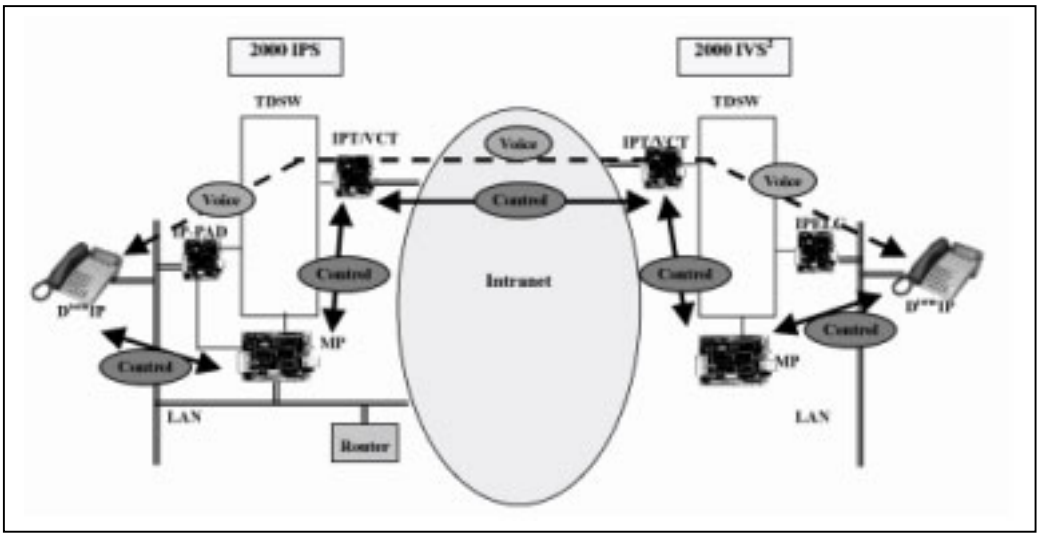


CCIS Networking via IP between Legacy Terminals



CCIS Networking via IP (Non Peer-to-Peer)

When the distant PBX system does not support peer-to-peer, the systems are connected with IP trunks (including Voice Compression Trunks (VCT)), via Time Division Switch (TDSW). Voice signals of IP terminals are transmitted via IP-PADs, while those of legacy terminals are directly connected to the IP trunks. Call control signals between the systems are also transmitted over the IP trunks. Voice compression of G.729a (8Kbps) and G.723.1 (5.3Kbps/6.3Kbps) can be provided by the IP trunks with VCT cards.



Service Conditions on Non Peer-to-Peer Connections

- IP trunk connections over CCIS Networking via IP are available when the distant systems are NEAX 2000 IVS^2, NEAX 2400 IMX/IPX or Electra Elite IPK that does not support peer-to-peer connections.
- The IP-PAD is required for connections between IP Enabled Dterm terminals and IP trunks.
- Maximum one IP trunk card can be accommodated per PIM. Maximum eight IP trunk cards can be accommodated in the system
- Maximum voice channels per IP trunk card depends on the payload period as follows (payload period can be assigned in system programming):

Payload Period	Maximum Voice Channels per IP Trunk
10 ms	4
20 ms	8
30 ms	16
40 ms	16

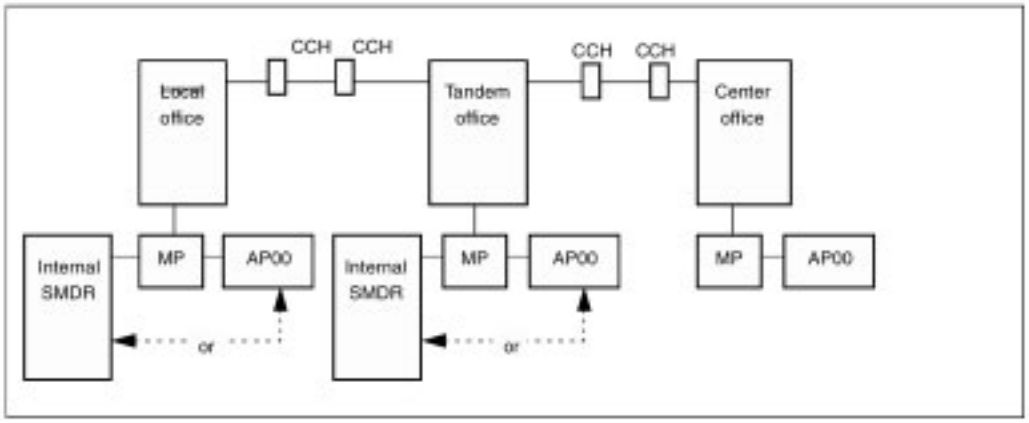
- Maximum 127 voice channels can be provided per system (point-to-multipoint connection).
- IP trunk cards must be connected with the intranet via router.

Centralized Billing

This feature is used to collect billing information from each office within the network and direct it to the associated center office. Billing information is then forwarded to the central billing centers via RS232C interfaces.

Service Conditions

1. The Centralized Billing system is composed of local office, tandem office, and center office.



2. The *UNIVERGE* NEAX 2000 IPS is generally used as a local office; however, it can be used as a center office for centralized billing if all the PBXs within the network are IPS.

3. When the *UNIVERGE* NEAX 2000 IPS is used as a center office, the billing information is sent to a SMDR system using the NEAX 2400 format and an AP00.

4. When a *UNIVERGE* NEAX 2400 IPX is used as a center office, either the Sub Line or the My Line number can be selected by system programming to be sent to the SMDR terminal. When the *UNIVERGE* NEAX 2000 IPS is used as a center office, only the My Line number can be sent to the SMDR terminal. When a call is originated from a station (Station A) of a local office (without AP00), via a COT of a center office (with AP00), a message is sent to the center office from the local office via CCIS. According to the station number of the calling party, the AP00 of the center office generates billing information to the SMDR terminal.

5. Maximum digits of station number is as follows:
   Local office/Tandem office:
   - using Internal SMDR: six digits (SMDR over IP is not supported)
   - using AP00 : four digits (if fifth digit of all stations uses a common leading digit then five digits)

   Center office:
   - using Internal SMDR: six digits
   - using AP00:four digits (if fifth digit of all stations uses a common leading digit then five digits)

6. Center office

   - Maximum of eight local offices can be accommodated.
   - Maximum of 3,600 calls per hour can be received. If the center office exceeds 3600, billing information is stored at local office.

7. Local office

   - By predetermined office data, local office number of the calling station can be forwarded to the center office, and output to the SMDR.

   - In case of using internal SMDR

     a. The billing information cannot be output at local office.
     b. When a failure of center office occurs or CCIS link is disconnected, the billing information is stored up to 1,024 calls at local office. System initializing loses stored information.
     c. Maximum of 3,600 calls per hour can be sent to center office/tandem office.

   - In case of using AP00

     a. Internal SMDR cannot be used.
     b. Maximum of 3,600 calls per hour can be sent to center office/tandem office.
     c. The billing information can be sent to center office/tandem office and at the same time it can be output at local office.

   - The billing information can be stored as follows and system initializing does not lose stored information.

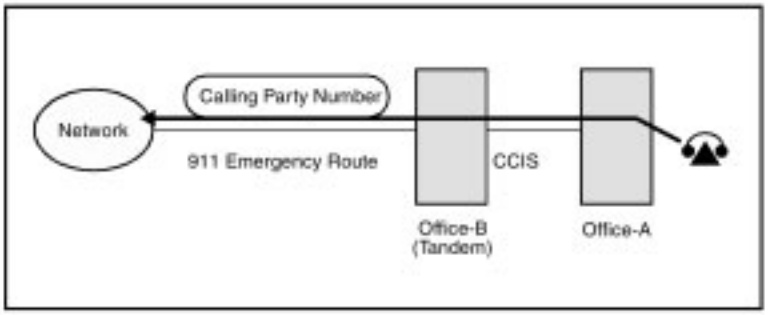
     a. AP00B MRC-C(AP) without expansion memory: up to 800 calls
     b. AP00B MRC-C(AP) with expansion memory: up to 27000 calls
     c. AP00B MRC-F(AP) without expansion memory: up to 1965 calls
     d. AP00B MRC-F(AP) with expansion memory: up to 22925 calls

8. Tandem Office

   - One center office and maximum of seven local offices can be accommodated.
   - Maximum of 3,600 calls per hour can be received from local office. If the center office exceeds 3600, billing information is stored at local office.
   - Maximum of 3,600 calls per hour can be sent to center office.
   - Tandem office also can function as a local office.

Centralized E911 – CCIS

This feature allows the system to transmit a calling party number to the 911 Emergency system over CCIS tandem connection.



Service Conditions

1. The Calling Party Information Transferring Service must be provided between calling office and tandem office.

2. The calling through CCIS must be Least Cost Routing (LCR) calling.

Call Set Up Times

Standard tie line networks typically require from two to eight seconds to complete the set up of a call and return ring back tone to the caller. With a CCIS network, call set up times are much lower averaging from 100 msec. to two seconds depending on the number of systems in the network.

Look Ahead Routing

In a standard non-CCIS network using tie lines to connect two nodes, when a station in node A calls a station in node B a tie line is seized and the call set up information is transmitted (in band) to attempt to ring down the called station. If that station is busy, busy tone is heard through the tie line from the destination system. This acceptable practice has been used for many years in spite of how inefficient it may be. The above scenario illustrates why many non-CCIS tie line networks require tie line usage for calls that never get completed. This is not very efficient and often results in customers needing to purchase more voice/data tie lines then what would really be needed if an intelligent network were employed.

The intelligent CCIS network provides a Look Ahead Routing feature that helps to reduce the load on network tie lines (voice/data channels) thereby reducing the total number of voice/data channels (tie lines) required between two nodes. When an on-net call is placed to a remote node, the system will “Look Ahead” to the destination system to determine if the called station is available. This is accomplished by communications between the systems over the D channel. If the called station is not available, busy tone is provided to the calling party locally and a voice/data channel (tie line) is not used. This provides a decrease in the overall number of tie lines needed to carry the traffic loads between network nodes, further reducing ongoing operation costs and system hardware requirements.

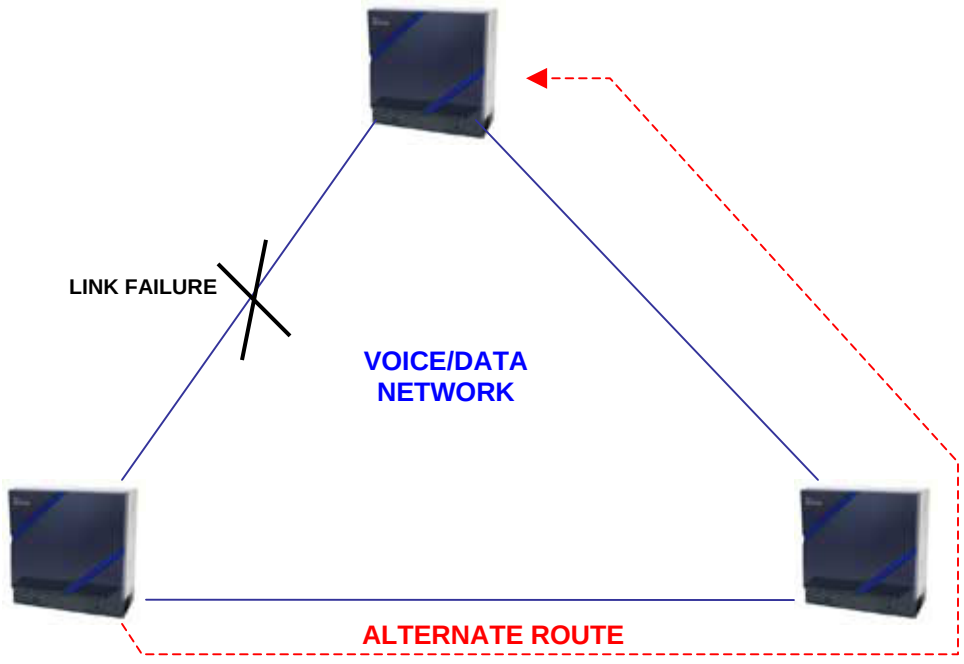
The benefits of Look Ahead Routing are also realized in applications where trunk facilities are strategically distributed throughout the network. CCIS will “Look Ahead” when a user in node A tries to access a trunk facility from node B. A tie line from node A to node B would only be used if the trunk facility requested is available.

Shared Trunk Facilities and Alternate Routing

Many network applications dictate a need to strategically distribute trunk facilities throughout the network. The practice of sharing trunk facilities across a network is beneficial because it reduces the overall network trunking requirements and provides additional lower cost route choices. When combining this practice with the Alternate Routing capabilities of CCIS, a large increase in network reliability and user productivity can be realized while reducing network operations costs.

The flexibility surrounding CCIS allows multiple alternate routing schemes, which can be created to enhance network reliability while they are specifically designed to consider Least Cost Routing (LCR) benefits. This provides the user with multiple alternate routes that are automatically selected with call costs as the primary factor in route selection. As shown in the following diagram, CCIS intelligence will also automatically use the pre-configured alternate route for the D channel (if configured for link redundancy) in the event that the primary D channel route is disabled.

Alternate Routing



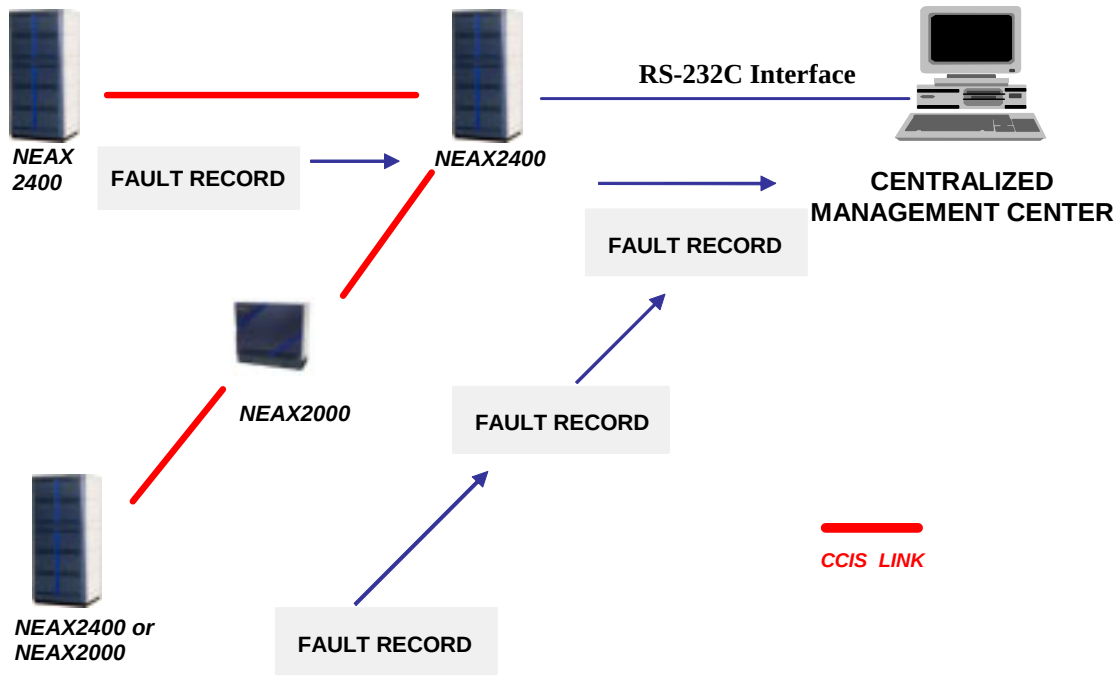
Centralized System Maintenance and Administration

A centralized system maintenance and administration facility can be designated at one node in the network allowing all nodes in the CCIS network to be maintained from a single location. This reduces travel costs and maintenance times. All system messages, fault data, and cyclic diagnostic reports will be transmitted over the D channel to the designated central facility as shown in the following diagram. The individual system messages and diagnostic data of each node can be easily identified, since each system node has an assigned name allowing messages from multiple sites to be separated. Additionally, each node will buffer and output system messages and diagnostic data to a local printer at the node generating the information.

This feature applies to the 2400s sharing a network with 2000s. The 2000 IPS will pass fault information originating from the 2400s through to other 2400s. When a *UNIVERGE* NEAX 2400 IPX is in the network, the 2000 IPS will send its fault data to the 2400 IPX designated as Central Fault MAT.

System administration also benefits from the ability to perform high speed adds, moves or changes from the centralized facility for any node in the network without the addition of costly peripheral equipment. The costs savings in this area are immediately realized.

Centralized Maintenance Facility

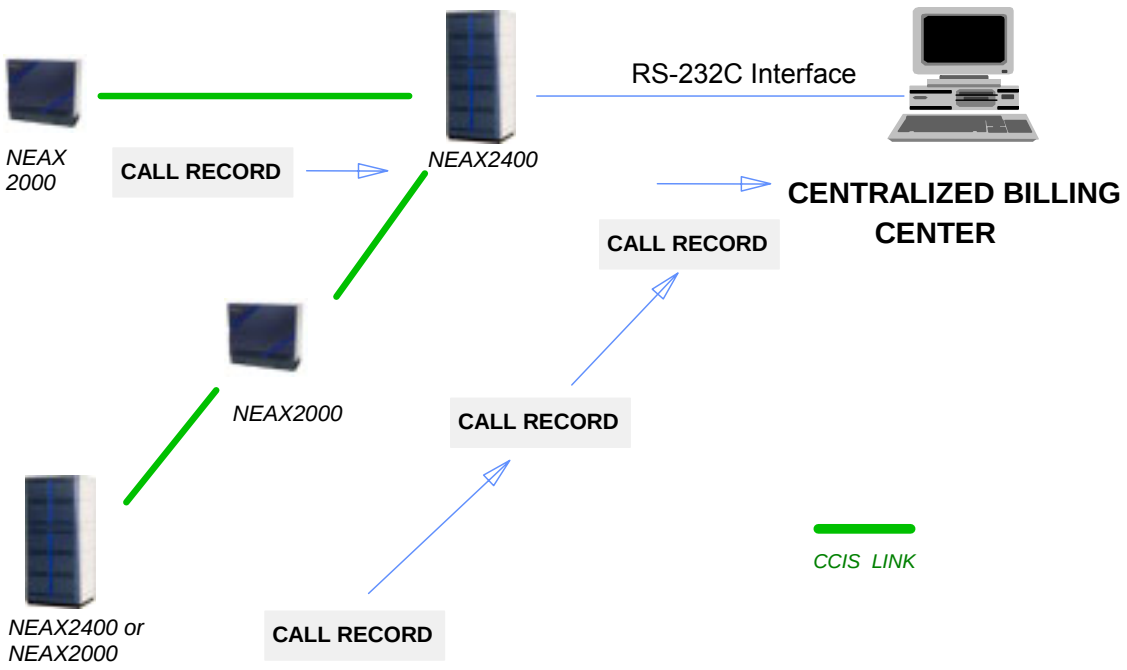


Centralized Call Accounting/Billing Systems

Standard tie line networks require independent call accounting / billing equipment for each node in the network. This results in a costly investment of peripheral equipment, higher maintenance costs, and the need to service this equipment at each node.

In a CCIS network, it is not necessary to purchase independent call accounting / billing equipment for each node in the network. One call accounting / billing system can be centrally located, connected to any one system in the network, and the intelligent CCIS network will transmit the SMDR data from each system in the network over the D-channel to the designated centralized collection point, as shown in the following diagram. This increases reliability by minimizing hardware requirements, simplifies call costing data analysis, and provides a large reduction in operation costs.

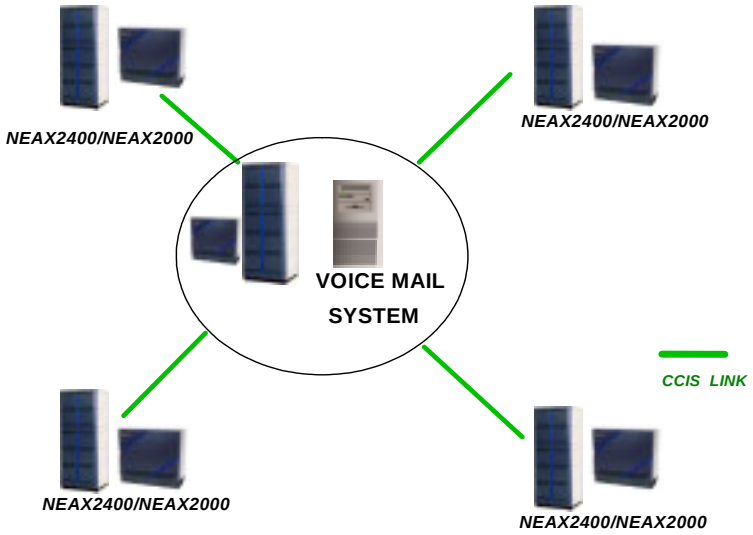
Centralized Call Accounting / Billing



Centralized Voice Processing / Messaging

Voice Processing Systems such as voice mail, fax mail, and Integrated Voice Response (IVR) systems can be centralized in a CCIS network. This eliminates the need for costly independent systems to be purchased and maintained for each node in the network. Maintenance is simplified by locating the system at the centralized maintenance facility or any other facility within the CCIS network. The information required for full integration from any of these types of systems will be transmitted over the D-channel to the required node as required by the integration logic of the voice processing system. Increased network reliability and substantial operating cost savings can be realized.

CCIS Centralized Voice Mail



Centralized Attendant Consoles

CCIS allows the centralization of attendant consoles to service calls from any system in the network. By centralizing attendant services, a greater number of calls can be handled with fewer consoles based on economies of scale. System hardware requirements can be reduced and overall operations costs are lower. CCIS uses the Look Ahead feature discussed earlier to minimize tie line usage as both on-net and off-net calls are centrally serviced and distributed throughout the network.

Uniform Numbering Plans

CCIS offers complete flexibility when designing network numbering plans. Two primary numbering schemes can be used.

One scheme uses a four- or five-digit station numbering plan with no system node identifier. In this scheme, station numbers are not repeated throughout the network and usually the first digit is used as an indicator of which node the station is at. For example; a three-node system might use station numbers 1000-2999 at node 1; 3000-4999 at node 2; and 5000-6999 at node 3. The intelligence of CCIS and LCR will automatically route the call to the correct node.

The second scheme uses a one, two or three digit node identification code followed by free station numbering. In this scheme station numbers can be repeated at each node in the network, if desired. This scheme is especially beneficial to those customers who are upgrading a network or who are networking multiple facilities for the first time and wish to maintain their existing numbering plans at each facility. It is also very useful in very large networks where it may be necessary to repeat station numbers at different facilities due to the total number of stations in the network.

CCIS Feature Chart

CCIS Feature Name	NEAX 2400 Required
Attendant Camp-On with Tone Indication-CCIS	
Attendant Controlled Conference-CCIS	YES
Automatic Recall-CCIS	
Brokerage-Hot Line-CCIS	
Busy Lamp Field (BLF)-CCIS	
Busy Verification-CCIS	
Call Back-CCIS	
Call Forwarding-All Calls-CCIS	
Call Forwarding-Busy Line-CCIS	
Call Forwarding-Don't Answer-CCIS	
Call Forwarding-Intercept-CCIS	
Call Forwarding-Override-CCIS	
Call Processing Indication-CCIS	
Call Transfer-All Calls-CCIS	
Call Transfer-Attendant-CCIS	
Calling Station Status Display-CCIS	
Calling Name Display-CCIS	
Calling Number Display-CCIS	
Call Transfer-All Calls-CCIS	
Centralized Billing-CCIS	
Centralized Day/Night Mode Change-CCIS	YES
Centralized E911-CCIS	
Consultation Hold-All Calls-CCIS	
Deluxe Traveling Class Mark-CCIS	
Dial Access to Attendant-CCIS	
Direct-in Termination-CCIS	

CCIS FEATURE CHART (Cont.)

CCIS Feature Name	NEAX 2400 Required
Distinctive Ringing-CCIS	
Do Not Disturb-CCIS	
Dual Hold-CCIS	
Elapsed Time Display-CCIS	
Flexible Numbering of Stations-CCIS	
Hands-Free-Answer Back-CCIS	
House-Phone-CCIS	
Hot Line-CCIS	
Incoming Call Identification-CCIS	
Individual Attendant Access-CCIS	
LDN Night Connection-CCIS	
Link Alarm Display-CCIS	
Link Reconnect-CCIS	
Message Waiting Lamp Setting-Attendant-CCIS	
Message Waiting Lamp Setting-Station-CCIS	
Miscellaneous Trunk Access-CCIS	
Miscellaneous Trunk Restriction-CCIS	
Multiple Call Forwarding-All Calls-CCIS	
Multiple Call Forwarding-Busy Line-CCIS	
Multiple Call Forwarding-Don't Answer-CCIS	
Multiple Console Operation-CCIS	
Night Connection Fixed-CCIS	
Night Connection Flexible-CCIS	
Outgoing Trunk Queuing-CCIS	
Paging Access-CCIS	
Restriction from Outgoing Calls-CCIS	
Service Display-CCIS	
Single Digit Station Calling-CCIS	
Station Controlled Conference-CCIS	YES
Station to Station Calling-CCIS	
Station to Station Calling-Operator Assistance-CCIS	
Toll Restriction-3/6 Digit-CCIS	
Trunk Answer from Any Station-CCIS	
Trunk to Trunk Restriction-CCIS	
Uniform Number Plan-CCIS	
Voice Call-CCIS	
Voice Mail Integration-CCIS	
Voice Mail Live Record-CCIS **Note**	
Voice Mail Private Password-CCIS	
Asynchronous Data Switching – CCIS	
Data Interface - Automatic Answer – CCIS	
Data Line Security – CCIS	
Data Transparency – CCIS	
Simultaneous Voice and Data Transmission – CCIS	
Synchronous Data Switching – CCIS	

Note: *Live Record Over CCIS is not supported to a NEAX 2400.*

System Capacity

System Capacity for CCIS with Digital Interface

Description	Capacity			
	24DTI	30DTI	24CCT	30CCT
DTI Card	8	8	—	—
CCH Card	8	8	—	—
AP00 Card (for Centralized Billing)	1	1	1	1
Trunks for DTI	192	248	—	—
CCIS Routes	8	8	8	8
Trunks per DTI Card	24	31	—	—
CCT Card	—	—	8	8
Trunks for CCT	—	—	200	248
Trunks per CCT Card	—	—	25	32
M10 Card	4	4	4	4

System Capacity for CCIS with Analog Interface

Description	Capacity
CCH Card	8
AP00 Card (for Centralized Billing)	1
ODT Card	128
ODT Trunks	254

System Capacity for CCIS Networking via IP

Description	Capacity
CCIS Trunks (IP Trunk/ Peer to Peer)	255
CCIS via IP (non Peer-to-Peer)	
SPN-IPTB	8
SPN-4VCTI	32
CCIS via IP (Peer-to-Peer)	
PZ-M606-A	1
SPN-8IPLA IP PAD-C	8
PZ-24IPLA	8

Required Equipment

Required Equipment for Digital CCIS

EQUIPMENT NAME	FUNCTIONAL NAME	FUNCTION
PN-24CCTA	CCT	CCIS (1.5 Mbps) Trunk Card Provides a built-in Common Channel Handler (CCH) of CCIS.
PN-24DTA-C	DTI	T1 Digital Trunk Interface (1.5 Mbps) Card Accommodates 24-channel PCM digital lines.
PN-30DTC-C	DTI	E1 Digital Trunk Interface (2 Mbps) Card Accommodates 30-channel PCM digital lines.
PN-SC00	CCH	Common Channel Handler Card Transmits/receives signals on the common signaling channel of CCIS.
PN-M10	M10	Optical Fiber Interface Card Provides optical fiber interface for T1/E1 Digital Trunk Interface (1.5 M/2 Mbps) Line length: 6.2 miles (10 km) or less Line coding: CMI

Required Equipment for Analog CCIS

EQUIPMENT NAME	FUNCTIONAL NAME	FUNCTION
PN-SC00	CCH	Common Channel Handler Card Transmits/receives signals on the common signaling channel of CCIS.
PN-2ODTA	ODT	2-line Out Band Dialing Trunk Card Used as either a 2-wire E&M trunk or a 4-wire E&M trunk. Equipped with –48V DC-DC on-board power supply. Both Circuits 0 and 1 must be set to the same purpose (2-wire or 4-wire) in one card.

Required Equipment for CCIS Networking via IP

Description	Remarks
CCIS via IP (non Peer-to-Peer)	
SPN-IPTB	Network Interface: ETHER/FAST ETHER
SPN-4VCTI	Codec Card for IP Trunk
CCIS via IP (Peer-to-Peer)	
PZ-M606-A	On board Ethernet Interface Card (ETHERNET TCP/IP)
SPN-8IPLA IP PAD-C	IP Interface card for IP Pad
PZ-24IPLA	24-channel daughter board for 8IPLA

IP Specifications

Item		Specifications	Remarks
Voice Encoding		G.729a G.723.1 G.711	8 Kbps CS-ACELP 5.3/6.3 Kbps MP-MLQ/ACELP 64 Kbps PCM
IP-PAD		8/32 channels per card Automatically seized per call	
FAX Communication Feature		FAX Relay Method (T.30)	G3 FAX (up to 14.4 Kbps) Super G3 Reciprocal: Not allowed FAX communication with H.323: Not available PN-32IPLA (IPPAD) card (PN-8IPLA (IP-PAD) card is not available.)
		FAX Relay Method (Pass-through (G.711/G.726))	PN-8IPLA (IP-PAD) card (PN-32IPLA (IPPAD) card is not available.)
DTMF Signal		H.245	H.323 IPT/IP-PAD/DtermIP
Inter-office/Intra-office Signaling		H.245	DtermIP-to-DtermIP connection DtermIP-to-IP-PAD connection
		PROTIMS over IP	DtermIP-to-2000 IPS connection
		CCIS over IP	Point-to-Multipoint connection
		H.323	H.323 IPT/4VCT card and IP PAD card are required
Jitter Control		Dynamic Jitter Buffer	
Quality of Service (QoS)		• TOS, IP Precedence • DiffServ	
LAN Interface		10BASE-T/100BASE-TX	Auto Negotiation is available. 100BASE-TX is recommended.
Echo Canceller (IP-PAD)		G.168 (64 ms.)	
Payload Size	DtermIP/ CCIS Virtual IPT (32IPLA + 16VCT)	10 ms.-40 ms. (G.723.1: 30 ms. fixed)	Maximum voice channels per card G.729a G.711 G.723.1 10 ms.: 12 ch 12ch – 20 ms.: 20 ch 20ch – 30 ms.: 30 ch 30ch 24ch 40 ms.: 32 ch 32ch –
	DtermIP/ CCIS Virtual IPT (8IPLA + 24IPLA)	10 ms.-40 ms. (G.723.1: 30 ms. fixed)	Maximum voice channels per card G.729a G.711 G.723.1 10 ms.: 20ch 20ch - 20 ms.: 32ch 32ch - 30 ms.: 32ch 32ch 24ch 40 ms.: 32ch 32ch -
	H.323 IPT	20 ms.-40 ms. (10 ms. increments) (G.723.1: 30 ms. fixed)	Maximum voice channels per card G.729a G.711 G.723.1 20 ms.: 6ch 5ch - 30 ms.: 8ch 7ch 8ch 40 ms.: 12ch 10ch -
PAD Control		0 dB to +16 dB (+2 dB increments) 0 dB to –16 dB (–2 dB increments)	Setting is available per Location No. For connection between DtermIPs
		0 dB to –16 dB (For North America) 0 dB to –12 dB (Outside North America)	For connection via the IPT card

DTI Specifications

Characteristics	24-Channel	30-Channel
Output Line Rate Line Code Line Impedance	1.544 Mbps – 50 ppm AMI with ZCS/B8ZS* 100 ohms	2.048 Mbps – 50 ppm HDB3 (High Density Bipolar 3) 75 ohms + 100 ㎐H (Coaxial Cable) 120 ohms + 160 ㎐H 2.37 volts nominal (Coaxial Cable) 3 volts nominal (Twisted-Pair Cable)
Pulse Amplitude (Base to Peak)	3 volts – 0.6 volts	
Pulse Width	324 ns – 30 ns	244 ns nominal
Input Line Rate Pulse Amplitude (Base to Peak)	1.544 Mbps – 2000 bps (130 ppm) 1.5 volts – 3 volts	2.048 Mbps – 50 ppm 1.5 volts – 2.7 (Coaxial Cable) 1.5 volts – 3.3 volts (Twisted-Pair Cable)
Frame Synchronization Pattern	100011011100	
Input Jitter	ITU-T Fig. 1/G743	ITU-T Fig. 1/G743
Wander	+138UI, -193UI or -138UI, +193UI	ITU G823
Cable Length from PBX to MDF or External Equipment	Max. 200 m (656.2 ft.) [With 0.65 gauge (22 ABAM) twisted pair cable]	Maximum 400 m (1312.4 ft.) [With 0.65 gauge (22 ABAM) twisted pair cable]

*AMI: Alternate Mark Inversion
ZCS: Zero Code Suppression
B8ZS: Bipolar Eight Zero Substitution.

Chapter 15 Integrated Services Data Network (ISDN)

ISDN Primary Rate Interface

ISDN is becoming a basic requirement for any PBX system sold today. The *UNIVERGE* NEAX 2000 IPS supports many ISDN BRI/PRI features and services. AT&T and Northern Telecom specifications are both supported by the IPS. ISDN PRI trunks can use all of the same features of the *UNIVERGE* NEAX 2000 IPS that any DID or DOD type trunk can use. For example, ISDN PRI trunks can terminate to the PBX and use features such as Internal Automated Attendant, DISA, Attendant Overflow, etc. In a typical installation, ISDN trunks will be terminated as DID type lines, but the IPS also allows ISDN trunks to terminate as ring down type lines. As this section will convey, the *UNIVERGE* NEAX 2000 IPS provides a very attractive ISDN PRI package.

ISDN FEATURE LIST
CALL-BY-CALL SERVICE SELECTION
CALLING PARTY RECOGNITION SERVICE (DIRECT-IN TERMINATION (DIT))
CPN TO NETWORK—PRESENT
CPN TO TERMINATING USER—DISPLAY
DID ADDRESSING
DID AND DOD ADDRESSING
EVENT BASED CCIS
ISDN TERMINAL
MEGACOM® ACCESS/WATS
MEGACOM® 800 SERVICE/800 WATS/ULTRA WATS
MULTIQUEST® /900 SERVICE
SUB-ADDRESS—PRESENT
TRUNK PROVISIONING SERVICE SELECTION

PRI Services & Features

Services provided by ISDN carriers are separate and different from features supported by ISDN carriers. Services, in general terms, are often entire packages provided by the carriers, such as AT&T's Megacom®, which provides routing options and many reporting features. Features of ISDN trunks provide the user with items such as Calling Party Number (CPN) and Sub-Addresses. The distinction between services and features should be recognized to a certain extent. This section will explain which features and services the *UNIVERGE* NEAX 2000 IPS supports.

Call-By-Call Service Selection

Services can be selected on a call-by-call basis to all channels of a single PRI interface according to applications. That is, unlike Trunk Provisioning Service in which services are assigned to specific channels, services may be used on any available channel.

Service Conditions

1. The services that can be designated include ACCUNET, MEGACOM, MEGACOM 800, INTERNATIONAL 800, SDN, MULTIQUEST (AT&T), and PRIVATE, INWATS, OUTWATS, FX, TIE (Northern Telecom).
2. Channel selection is possible by the LCR function only.
   Note: *During call termination, there is no indication of which service is being used.*
3. Up to 5-digit Network ID of Network-Specific Facilities Information Element can be sent to ISDN Network.
4. Transit Network Selection Information Element can be sent to ISDN Network.
5. Service conditions #3 and #4 are available with the following firmware of PN-24PRTA:
   SC-2795 IXS PRT PROG-A1: Issue 2.01 or later
   SC-2888 IXS PRT PROG-A2: Issue 1.01 or later
   When the firmware issue is earlier than above, up to 3-digit Network ID can be sent to the ISDN Network.

Called Party Recognition Service (Direct-In Termination (DIT))

This feature provides an incoming *Direct-In Termination* (DIT) call via an ISDN trunk to be connected to a predetermined station. This application can be used for a station or modem. For incoming calls in a Primary Rate Interface trunk, this service feature should be used only when DID trunks are not desired.

Note*:* *Refer to the Business Features and Specifications manual for details on service Conditions*

CPN To Network–Present

This feature allows the ISDN network to be informed of the calling party number (CPN) when a call originates from a terminal connected to the System.

Service Conditions

1. The data provided as the calling station number is assigned via MAT/CAT for each ISDN circuit or station. In addition, if no data has been assigned as the calling station number, the system will not provide any information to the network.
2. A maximum of 16 digits can be assigned as the calling station number.
3. The maximum number of area codes and office code patterns is 15.
4. Transmission of CPN to the ISDN network can be programmed by the Class of Service. Programming is required for each station.
5. The delivery of CPN information is subject to local regulations.

CPN To Terminating User–Display

This feature provides a visual display of the originating station's number and subaddress information on a Multiline Terminal or an Attendant Console for incoming ISDN calls. This provides the terminal user with a quick and accurate way to identify the originating station's number (Calling Party Number).

Service Conditions

1. This feature is available on the Multiline Terminal with Display, Attendant Console and Dterm PSIII.
2. A maximum of 16 digits forming the originating party's number (CPN), including the PBX access code, can be displayed. For sub-address, a maximum of 8 digits can be displayed. If the subaddress exceeds 8 digits, the first 8 are displayed.
3. The CPN is flashing while the Multiline Terminal is ringing on its Prime Line. The duration of display after the call is answered can be selected by system data programming (display for 6 seconds or continuously).
4. If the ISDN network provides Name Display service (NI-2/DMS100), the calling party name can be displayed in place of the CPN (A maximum of 16 characters).
5. The calling name from ISDN can be received by terminating systems (CM30 YY=02, 03) such as Trunk Direct Appearance (02), Trunk Direct Appearance + TAS (03), Direct-In Termination (04), Automated Attendant(09), ISDN Indial (18), and Attendant Console (14). In addition, Sub-address termination is available.
6. The 24PRT card is required.
7. When Call Forwarding or Call Transfer routes an incoming call from ISDN to a Multiline Terminal/PS in the own office, the calling name is displayed on the destination station of Call Forwarding or Call Transfer.
8. Calling Name Display on ISDN Terminals is not supported.
9. Calling Name Display or Calling Number Display can be selected as the initial display in station Class of Service.
10. The display can be changed to the calling name or calling number by pressing the display selection key on a Multiline Terminal or an Attendant Console while the number or name is displayed.
11. When the terminating system (CM30 YY=02, 03) is set to Trunk Direct Appearance or Attendant Console, the calling name is displayed only after the called party answers the call.
12. Conditions on Tandem Connection
    a. The calling name received from ISDN is relayed, only via CCIS, with tandem connection. (A maximum of 16 characters)
    b. When Call forwarding routes an incoming call from ISDN to CCIS, the calling name is relayed via CCIS to the destination of Call Forwarding.
    c. When the calling name from NI-2 is received as a FACILITY message, the name cannot be relayed to CCIS. Only when the calling name from NI-2 is received as a SETUP message, the name can be relayed to CCIS.
    d. Only when the calling name from DMS100 is received as a SETUP message, the name can be relayed to CCIS.
13. Conditions with Other Services
    a. The calling name received from ISDN has priority over the calling name registered to Station Speed Dialing by CM74, so the former is displayed even though the latter has been assigned by CM74.
    b. If the calling name is not received from ISDN, the calling name assigned by CM74 is displayed if available.
    c. When a call is relayed to CCIS with tandem connection, the calling name is not relayed to CCIS, if the name is not received from NI-2, even though the calling name has been assigned by CM74. However, when Call Forwarding - No Answer routes an incoming call from ISDN to CCIS with tandem connection, the calling name assigned by CM74 is relayed to CCIS.

DID Addressing

This feature allows incoming ISDN-PRI calls to terminate to stations, Attendant Console, Automated Attendant, etc., based on the Called Party number. Direct Inward Dial trunks will be terminated to preprogrammed destinations without Attendant assistance.

Service Conditions

1. If the network is notified that the destination station for the DID call is busy or a connection-controlled station, the network gives the calling station a Busy Tone. (This depends upon call forwarding services being in service.)
2. If the called station is nonexistent, the DID call can be routed to the Attendant Console, another pre-designated station, or to receive Reorder Tone.

Note: *Refer to the Business Features and Specifications manual for more details of DID service.*

DID and DOD Addressing

This feature allows the system to use DID and DOD on the same B channels. Trunk Provisioning Service Selection is not required. (B-channels can be used for DID and DOD without separating the trunk routes.)

Service Conditions

1. Confirm the DID and DOD selection with the customer's local exchange carrier prior to installation.
2. The DID/DOD is supported without service provisioning, since no facility indication to the network is required.

Note: *For more details, refer to "Direct Inward Dialing" and "Direct Outward Dialing" in the Business Features and Specifications manual*.

ISDN Terminal

This feature provides the system with an ISDN Terminal or Terminal Adapter (TA). ISDN Terminal to ISDN Terminal, ISDN Terminal to ISDN Trunk, ISDN Trunk to ISDN Terminal, ISDN Terminal to Single Line Telephone, ISDN Terminal to Multiline Terminal, and ISDN Terminal to PS connections are available.

Service Conditions

1. The ILC (ISDN Line Circuit) card and the ICH (ISDN Channel handler) card are required.
2. The ISDN Terminal must be locally powered.
3. The following connections are available:
   - Point to Point connection
   - Point to Multipoint connection
   - National ISDN 1
   - AT&T
4. The following features are available:
   - Individual Terminal Calling (Point to Multipoint connection)
   - Group Calling (Point to Multipoint connection)
   - Called Party Recognition Service (DIT)
   - DID Addressing
   - DID and DOD Addressing
   - CPN to Network-Present
   - CPN to Terminating User-Display
   - Sub Address-Present
   - Direct Outward Dialing (DOD)
   - Restriction from outgoing call
   - Toll restriction

- Station to Station Calling
- Station Hunting
- Simultaneous voice and data transmission
- Asynchronous Data Switching
- Synchronous Data Switching
- Half/Full Duplex Switchover

5. The ISDN Terminal can provide Voice and data communication via the CCIS network. The CCIS network must be digital.
6. During communication via the CCIS network, the link reconnection is not available.
7. The conditions about station services are as follows:
   - Station Hunting
     When a Single Line Telephone/Multiline Terminal or an ISDN Terminal calls a busy Single Line Telephone/Multiline Terminal, another Single Line Telephone/Multiline Terminal in Station Hunting group is called. When a Single Line Telephone/Multiline Terminal or an ISDN Terminal calls a busy ISDN Terminal, an-other ISDN Terminal in Station Hunting group is called.
     Note: *An ISDN Terminal cannot be included in the same Station Hunting group as Single Line Telephone/Multi-line Terminals.*

   - Call Forwarding
     An ISDN Terminal cannot set Call Forwarding - All Calls/- Busy Line/-Don't Answer, and cannot be the destination of a Call Forwarding. When an ISDN Terminal is a calling station, it can be forwarded to another Single Line Telephone/Mul-tiline Terminal but cannot be forwarded to central office trunk or tie line trunk.

   - Call Transfer
     A Single Line Telephone/Multiline Terminal cannot transfer a call in progress with an ISDN Terminal to another station. In the same way, an ISDN Terminal cannot transfer a call in progress with a Single Line Telephone/Mul-tiline Terminal to another station. While a Single Line Telephone/Multiline Terminal converses with CCIS trunk and ISDN trunk, the Single Line Telephone/Multiline Terminal cannot transfer the call to an ISDN Terminal.

   - Call Pickup, Call Pickup - Designated group
     An ISDN Terminal cannot be assigned to Call Pickup group and cannot pickup a call to another station. A Single Line Telephone/Multiline Terminal cannot pickup a call to an ISDN Terminal.

   - Executive Override
     During voice communication between a Single Line Telephone/Multiline Terminal and an ISDN Terminal, another Single Line Telephone/Multiline Terminal can interrupt into only Single Line Telephone/Multiline Terminal. During data communication with an ISDN Terminal, Executive Override is not allowed.

   - Busy Service (Step Call, Call Back)
     When a Single Line Telephone/Multiline Terminal calls a busy ISDN Terminal, busy service such as Step Call and Call Back cannot be provided to the ISDN Terminal. It is the same when an ISDN Terminal calls a busy Single Line Telephone/Multiline Terminal.

8. When a Single Line Telephone is calling an ISDN Terminal or an ISDN Terminal is busy, hooking service is not available. Therefore, at this time other services are not available.
9. A Multiline Terminal can call an ISDN Terminal via Primary Extension or Sub Line. And a Multiline Terminal can be called from an ISDN Terminal via Primary Extension or Sub Line.
10. When a Single Line Telephone/Multiline Terminal calls an ISDN Terminal, it can send the calling station number to the ISDN Terminal. At this time, ISDN trunk number and local number are also attached to be sent.
11. Station to Station calling between an attendant console and an ISDN Terminal is not available.

12. In case of Point to Multipoint connection, a Single Line Telephone/Multiline Terminal must dial ISDN Multi-point station number assigned by CM1B, not the ISDN Terminal number assigned by CM10.
13. When a Multiline Terminal calls an ISDN Terminal or an ISDN Terminal calls a Multiline Terminal, a calling station number is displayed to the calling Multiline Terminal or the ISDN Terminal. In case of Point to Multipoint connection, the calling station number displayed is the ISDN station number assigned by CM10.
14. Station to Station calling between a PS station and an ISDN Terminal is available.
15. Only Preset Dialing can be used from an ISDN Terminal, Overlap dialing is not available.
16. For E-CCIS you must have 64K unrestricted digital data connections (circuit mode). The Telephone Company cannot overflow these ISDN calls to analog lines.
17. Group Call, This feature terminates a call to all ISDN Terminals or Terminal Adapters accommodated on the same bus. Group Call is available for following connections:
    - ISDN trunk to ISDN Terminal
    - ISDN Terminal to ISDN Terminal
    - Single Line Telephone to ISDN Terminal (Series 3200 R6.2 enhancement)
    - Multiline Terminal to ISDN Terminal (Series 3200 R6.2 enhancement)

MEGACOM ® Access/WATS

AT&T's MEGACOM ® (WATS) network, as well as WATS from other carriers, can be used.

Service Conditions

1. The available WATS service is limited to MSB (Maximal Subscribed WATS Band).
2. No specific band will be indicated to the Network.

MEGACOM ® 800 Service/800 WATS Ultra WATS

AT&T's MEGACOM ® 800 (Inward WATS) network, as well as 800 WATS provided by other carriers, can be used.

Service Conditions

1. Command 76 can be used to convert indialed digits.
2. The Dialed Number Identification Service (DNIS) must match the station numbering plan (CM20).
3. Multiple DNIS are supported.

MULTIQUEST ® /900 Service

AT&T's MultiQuest ® service can be used. (It is a "900"-type service.) Also, 900 service provided by other carriers can be used.

Service Conditions

1. The DNIS (Dialed Number Identification Service) must match the numbering plan (CM20) of the 2000 IPS.
2. Multiple DNIS numbers are supported.
3. Command CM76 can be used to convert DNIS digits received from the ISDN Network.

Subaddress–Present

This feature allows a primary rate interface ISDN trunk to transfer the called party subaddress information to a destination ISDN station when the call is originated by the system. Dialing the called party station number and subaddress is required.

Service Conditions

1. If the calling party fails to dial the called party subaddress, ISDN cannot transfer any called party subaddress information to the destination party.

2. If a calling party does not wish to provide a called party subaddress, the call must terminate with the # key (Immediate Start). If the # key is not entered, a Timing Start operation begins. The Timing Start uses the interdigit timeout operation.
3. Subaddress dialing is available only on those telephone terminals that can generate push-button (DTMF) signals.
4. The called party subaddress must not exceed 8 digits.
5. The called party subaddress can be sent with trunk direct dial access.
6. This feature cannot be used when a call is originated to ISDN using Speed Dialing or Call Forwarding features.

Trunk Provisioning Service Selection

Each channel of a PRI interface can be dedicated to a particular service. Services are designated to specific channels; once designated, a channel can be used only for that service.

Service Conditions

1. The services that can be designated include MEGACOM ® /WATS and MEGACOM ® 800/800WATS.
2. Arrangement with the carrier at the time of provisioning is needed.

Event Based CCIS

This feature allows a PBX customer that does not have a tie line (or when a customer cannot use the tie line due to the busy or fault of the line), to use the various CCIS feature by using the ISDN line as a CCIS virtual tie line, on the 2400 IPX - 2000 IPS connection or the 2000 IPS - 2000 IPS connection.

Service Conditions

1. Event Based CCIS is available between 2000 IPS to 2400 IPX and 2000 IPS to 2000 IPS.
2. The maximum number of virtual tie lines is 16 channels per system. This includes the common signaling channels and voice channels.
3. The ISDN line used for the virtual tie line can also be used as a regular ISDN line.
4. This feature supports voice calls only. (Supported object at PBX transmission side: Single Line Telephone, Multiline Terminal, DID/E&M/Ring Down (analog/T1) tandem calls). The data calls are transmitted via the regular ISDN network.
5. The Peg Count (the number of originating call from the ISDN trunk) is counted when using the ISDN line for the virtual tie line by route basis.
6. Billing information of the virtual tie line using the ISDN line is treated as regular tie line calls.
7. Billing information of the virtual tie line using the ISDN line is treated on tandem calls.
8. The voice channel of the virtual tie line is released at a programmable time after the call finishes.
9. The CCH (Common Channel Handler) card is required for providing Event Based CCIS.
10. For Event Based CCIS using PRI, the following services must be ordered from Telco on the PRI Span:
    - Each B channel must be capable of making and receiving 64K voice calls.
    - Each B channel must be capable of making and receiving 64k Unrestricted Digital Data calls.
    - The ISDN PRI span must be ordered with ISDN Indial (DID) service.
11. The PRI trunks should be ordered with the following components included:
    - Protocol must be National ISDN-1
    - Calling and Called Party Sub-Address
    - NIUF Capability Package S
    - Two B-channels capable of Voice and Data simultaneously on both B-channels
    - Two Directory numbers, each with a SPID
12. For any feature that requires a tandem connection (Call-Forward CCIS, etc.), additional Telco BRI circuits are required. More than one BRI trunk is required to allow calls to go back to the PBX that transferred or forwarded a call.

Business Feature List

Business Feature List	
Account Code	Elapse Call Timer
Answer Key	Forced Account Code
Attendant Delay Announcement	Group Listening
Attendant Overflow	Hold - Call
Authorization Code	Hold - Exclusive
Automated Attendant	Hold - Nonexclusive
Broker's Call	Last Number Redial
Call Forwarding	Least Cost Routing 3/6 Digit
Call Forwarding - All	Maintenance Administration Terminal(MAT)
Calls Call Forwarding - Busy Line	Fault Messages
Call Forwarding - No Answer	Peg Count
Call Forwarding - Outside	Night Service
Multiple Call Forwarding - All	Call Rerouting
Multiple Call Forwarding - Busy	Day/Night Mode Change - ATTCON
Multiple Call Forwarding - No Answer	Day/Night Mode Change - Station
Split Call Forwarding - Busy Line	Night Connection - Fixed
Call Forwarding - Override	Night Connection - Flexible
Group Division	Trunk Answer from Any Station(TAS)
Call Park - System	Route Advance
Call Pickup - Direct	Save and Repeat
Call Pickup - Group	Stack Dialing
Class of Service	Station Hunting - Circular
Code Restriction	Station Hunting - Terminal
Conference	Station Hunting - Secretarial
Consecutive Speed Dial	Station Message Detail Recording(SMDR)
Consultation Hold	Station Speed Dialing
Delayed Ringing	System Speed Dialing
Direct Inward Termination(DID)	Tenant Service
DID Call Waiting	Trunk Queuing - Outgoing
DID Digit Conversion	Trunk to Trunk Connection
Direct Inward System Access(DISA)	Uniform Call Distribution(UCD)
Direct Inward Termination(DIT)	"Silent Monitor"
Distinctive Ringing	Voice Mail Integration
Do Not Disturb	

ISDN Network Requirements for Layer One (1)

- Interface: T1 (1.544 M) only
- Framing: ESF (24 Multi-frame) only
- Line Coding: B8ZS only
- D-Channel Rate: 64 kbps only
- D-Channel: T1 Channel 24 (24th channel)

Note: *Individual B-Channels cannot be looped back, local or remote*.

Supported Network Services (Trunk provisioned only)

AT&T

- Software Defined Network
- Global Software Defined Network
- International 800 Service
- International MEGACOM
- AT&T's Alternate Destination Call Redirection feature is supported by the IPS.

NT

- Incoming **Note**
  - Private
  -Tie
  - WATS
  - 800 WATS

- Outgoing **Note**
  - DID/DOD

Note: *Services that do not require NSF can be supported by NEAX 2000 IPS.*

ISDN PRI Specifications

DTI

The Digital Trunk Interface (DTI) interfaces the PBX directly to 24/30-channel PCM transmission line. The DTI has the following functions.

For 24DTI:

- Two Directory numbers, each with a SPID
- Unipolar/Bipolar Conversion (AMI/B8ZS Format)
- Alarm Detection/Insertion
- Digital PAD on Voice Signal Transmission
- Loop-Back Test (Local/Remote Loop Back)
- Cyclic Redundancy Checking (based on ITU-T Rec. G704)

For 30DTI:

- Unipolar/Bipolar Conversion (HDB3 Format)
- Alarm Detection/Insertion
- Digital PAD on Voice Signal Transmission
- Loop-Back Test (Local/Remote Loop Back)
- Cyclic Redundancy Checking (based on ITU-T Rec. G704)

For connections of 24DTI and transmission line, twisted-pair cable can be used. For connection of 30DTI and transmission line, either coaxial cable or twisted pair cable can be used.

DCH

The D-Channel Handler (DCH) provides the D-Channel signaling interface through the DTI to an ISDN exchange, and it is responsible for signaling between the PBX and the ISDN exchange under control of the system MP.

PRT

The Primary Rate Interface Trunk (PRT) provides the ISDN Primary Rate Interface (1.5 Mbps PCM-23B + D/2 Mbps PCM-30B + D) and a built-in D-Channel Handler (DCH). The PRT has the following functions.

For 24PRT:

- Unipolar/Bipolar Conversion (AMI/B8ZS Format)
- Alarm Detection/Insertion
- Digital PAD on Voice Signal Transmission
- Loop-Back Test (Local/Remote Loop Back)
- Cyclic Redundancy Checking (based on ITU-T Rec. G704)

For 30PRT:

- Unipolar/Bipolar Conversion (HDB3 Format)
- Alarm Detection/Insertion
- Digital PAD on Voice Signal Transmission
- Loop-Back Test (Local/Remote Loop Back)
- Cyclic Redundancy Checking (based on ITU-T Rec. G704)

For connections of 24PRT and transmission line, twisted-pair cable can be used. For connection of 30PRT and transmission line, either coaxial cable or twisted pair cable can be used.

Note: *ISDN requires B8ZS Line coding with Extended Superframe (ESF) format*.

BRT

The Basic Rate Interface Trunk (BRT) provides one or two physical interface to the ISDN-Basic Rate Interface service (192 kbps PCM-2B + D).

The BRT has the following functions.

- Unipolar/Bipolar Conversion (AMI/B8ZS Format) (S/T Interface)
- Signaling Insertion/Extraction
- Frame Synchronization
- Digital PAD on Voice Signal Transmission

For connections of BRT and transmission line, twisted-pair cables can be used.

Note: *We recommend the point-to-point connection when connecting the system to the public network using the BRT card (Set the second data of CM35 Y=79 to 0). For the point-to-multipoint connection using the BRT card, when the system is established far from the public network, a communication error occurs easily because the ISDN signal fades away*.

Feature Compatibility Table

Inter-Exchange Carrier Network Voice Services and Features / *UNIVERGE* NEAX 2000 IPS Compatibility						
SERVICES AND FEATURES	**INTER-EXCHANGE CARRIERS**					
CALL BY CALL SERVICE SELECTION	**AT&T 4ESS**	**AT&T NET**	**NTI DMS 250**	**MCI NET**	**SPRINT NET**	***UNIVERGE* NEAX 2000 IPS**
Basic Call **Note 1**			X	**Note 1**	**Note 1**	**Note 1**
WATS	X	X	X	X	X	X
800 Service	X	X	X	X	X	X
Virtual Private Network	X	X	X	X	X	–
900 Service **Note 2**	X	X	X	X	–	X
900 Service Flexible Billing	X	X	–	–	–	–
International WATS	X	X	X	X	X	X
International 800 Service	X	X	X	X	X	X
Foreign Exchange (FX)	–	–	X	X	X	X
Tie Line	–	–	X	X	X	X
DEDICATED ACCESS PROVISIONING	**AT&T 4ESS**	**AT&T NET**	**NTI DMS 250**	**MCI NET**	**SPRINT NET**	***UNIVERGE* NEAX 2000 IPS**
Basic Call **Note 1**	–	–	X	**Note 1**	**Note 1**	**Note 1**
WATS	X	X	X	X	X	X
800 Service	X	X	X	X	X	X
Virtual Private Network	X	X	X	X	X	X
900 Service (**Note 2**)	X	X	X	X	–	X
900 Service Flexible Billing	X	X	–	–	–	–
International WATS	X	X	X	X	X	X
International 800 Service	X	X	X	X	X	X
Foreign Exchange (FX)	–	–	X	X	X	X
Tie Line	–	–	X	X	X	–
FACILITY MANAGEMENT	**AT&T 4ESS**	**AT&T NET**	**NTI DMS 250**	**MCI NET**	**SPRINT NET**	***UNIVERGE* NEAX 2000 IPS**
B-Channel Negotiation	X	X	–	–	–	–
D-Channel Back-Up	X	X	X	X	X	–
Non-Facility Associated Signaling	X	X	X	X	X	–
Release Link Trunks	–	–	X	X	X	–

X = Feature Available / – = NOT Available

VOICE FEATURES	AT&T 4ESS	AT&T NET	NTI DMS 250	MCI NET	SPRINT NET	*UNIVERGE* NEAX 2000 IPS
CPN to Network	X	X	X	X	X	X
Presentation Restrict/Allow	X	X	X	X	X	X
CPN/BPN Delivery	X	X	X	X	X	X
Provisioned by Service	X	X	X	X	X	X
Call by Call Request	X	X	X	X	X	-
While All Trunks Busy	-	-	X	X	X	-
CPN Sub-Address	X	X	X	X	X	X
Calling Party Name	-	-	X	X	X	X
Dialed Number ID Service (DNIS)	X	X	X	X	X	X
Original CPN	-	-	X	X	X	-
Original CPName	-	-	X	X	X	-
Redirecting Number	X	X	X	X	X	-
Redirecting Name	-	-	X	X	X	-
Redirection Number	-	-	X	X	X	-
Redirection Name	-	-	X	X	X	-
Called Party Sub-Address	X	X	-	-	-	X
Connected Party Number	X	X	X	X	X	-
Connected Party Name	-	-	X	X	X	-
User To User	-	-	-	-	-	-
Message Associated UUI	X	X	-	-	-	-
Call Associated TSC	X	X	-	-	-	-
Non-Call Associated TSC	X	X	X	X	X	-
Network Ring Again	-	-	X	X	X	-
Customer Group Info. Transport	-	-	X	X	X	-
Alternate Destination	X	X	-	-	-	-
Account Code Prompt and Send	X	X	X	X	X	-

X = Feature Available / - = NOT Available

Note 1: *The use of a Basic Call type of call is unique to NTI in the direct connection to the long distance network. This is sometimes referred to by the IEX carriers as "Megalink" service.*

Note 2: *AT&T and NTI have used different signaling to indicate the "900 service" selection. The NEAX2000 IVS supports the AT&T defined signaling.*

Local Exchange Carrier Network Voice Services and Features/ *UNIVERGE* NEAX 2000 IPS Compatibility						
SERVICES AND FEATURES	**INTER-EXCHANGE CARRIERS**					
CALL BY CALL SERVICE SELECTION SWITCHED DATA	**AT&T 4ESS**	**AT&T NET**	**NTI DMS 250**	**MCI NET**	**SPRINT NET**	***UNIVERGE* NEAX 2000 IPS**
64 kbps Clear Channel	X	X	X	X	X	X
64 kbps Restricted	X	X	X	X	X	–
56 kbps Restricted	X	X	X	X	X	–
International 64 kbps Clear	X	X	X	X	X	X
International 64 kbps Restricted	X	X	X	X	X	–
International 56 kbps Restricted	X	X	X	X	X	–
HO 384 kbps	X	X	X	X	–	–
H11 1526 kbps	X	X	X	X	–	–
N x 64 kbps	–	–	X	X	–	–
VIRTUAL PRIVATE NETWORK	**AT&T 4ESS**	**AT&T NET**	**NTI DMS 250**	**MCI NET**	**SPRINT NET**	***UNIVERGE* NEAX 2000 IPS**
64 kbps Clear Channel	X	X	X	X	X	–
64 kbps Restricted	X	X	X	X	X	–
56 kbps Restricted	X	X	X	X	X	–
International 64 kbps Clear	X	X	X	X	X	–
International 64 kbps Restricted	X	X	X	X	X	–
International 56 kbps Restricted	X	X	X	X	X	–
HO 384 kbps	X	X	X	X	–	–
H11 1526 kbps	X	X	X	X	–	–
N x 64 kbps	–	–	X	X	–	–
DEDICATED ACCESS PROVISIONING SWITCHED DATA	**AT&T 4ESS**	**AT&T NET**	**NTI DMS 250**	**MCI NET**	**SPRINT NET**	***UNIVERGE* NEAX 2000 IPS**
64 kbps Clear Channel	X	X	X	X	X	X
64 kbps Restricted	X	X	X	X	X	–
56 kbps Restricted	X	X	X	X	X	–
International 64 kbps Clear	X	X	X	X	X	X
International 64 kbps Restricted	X	X	X	X	X	–
International 56 kbps Restricted	X	X	X	X	X	–
HO 384 kbps	X	X	X	X	–	–
H11 1526 kbps	X	X	X	X	–	–
N x 64 kbps	–	–	X	X	–	–

X = Feature Available / – = NOT Available

VIRTUAL PRIVATE NETWORK	AT&T 4ESS	AT&T NET	NTI DMS 250	MCI NET	SPRINT NET	*UNIVERGE* NEAX 2000 IPS
64 kbps Clear Channel	X	X	X	X	X	–
64 kbps Restricted	X	X	X	X	X	–
56 kbps Restricted	X	X	X	X	X	–
International 64 kbps Clear	X	X	X	X	X	–
International 64 kbps Restricted	X	X	X	X	X	–
International 56 kbps Restricted	X	X	X	X	X	–
HO 384 kbps	X	X	X	X	–	–
H11 1526 kbps	X	X	X	X	–	–
N x 64 kbps	–	–	X	X	–	–

Local Exchange Carrier Network Voice Services and Features *UNIVERGE* NEAX 2000 IPS Compatibility				
SERVICES AND FEATURES	**LEC CARRIERS**			**UNIVERGE NEAX 2000 IPS**
CALL BY CALL SERVICE SELECTION	**AT&T 5ESS**	**NTI DMS100**	**SIEMENS EWSD**	
Basic Call	X	X	X	X
WATS				
Maximal Band	X	X	X	X
Banded	X	X	X	–
800 Service (INWATS)	X	X	X	X
FX LINE	–	X	–	–
TIE LINE	–	X	–	–
Virtual Private Network	X	X	–	X
Electronic Tandem Network	X	–	–	–
Operator Access (Note 2)				
LEC Operator	X	X	X	X
Default IEX Operator	X	X	X	X
Other IEX Operator	–	X	–	–
DEDICATED ACCESS PROVISIONING	**AT&T 5ESS**	**NTI DMS100**	**SIEMENS EWSD**	**UNIVERGE NEAX 2000 IPS**
Basic Call	X	X	X	X
WATS				
Maximal Band	X	X	–	X
800 Service (INWATS)	X	X	–	X
FX LINE	–	X	–	–
TIE LINE	–	X	–	–
Virtual Private Network	X	X	–	X
Electronic Tandem Network	X	–	–	–

X = Feature Available / – = NOT Available

	AT&T 5ESS	NTI DMS100	SIEMENS EWSD	*UNIVERGE* NEAX 2000 IPS
Operator Access (Note 2)				
LEC Operator	X	X	-	X
Default IEX Operator	X	X	-	X
Other IEX Operator	-	X	-	-
	-	-	-	-
FACILITY MANAGEMENT				
D-Channel Back-Up	-	X	-	-
Non-Facility Associated Signaling	-	X	-	-
VOICE FEATURES	**AT&T 5ESS**	**NTI DMS100**	**SIEMENS EWSD**	***UNIVERGE* NEAX 2000 IPS**
CPN to Network	X	X	X	X
Presentation Restrict/Allow	X	X	X	X
CPN/BPN Delivery	X	X	X	X
Calling Party Name	-	X	-	-
Original CPN	-	X	-	-
Original CPName	-	X	-	-
Redirecting Number	X	X	-	-
Redirecting Name	-	X	-	-
Redirection Number	-	X	-	-
Redirection Name	-	X	-	-
Connected Party Number	X	X	-	-
Connected Party Name	-	X	-	-
User To User	-	-	-	-
Message Associated UUI	X	-	X	-
Network Ring Again	-	X	-	-
Network Message Waiting				
Message Waiting Indicator	-	X	-	-
Message Waiting Activation	-	X	-	-

X = Feature Available / - = NOT Available

Note 1: *The Siemens EWSD offers PRI support via both a National ISDN (Bellcore) and an AT&T "Custom" implementation. The above information pertains to the "Custom" implementation only (NTI has no National ISDN support). Although this interface is based on the AT&T 5ESS, NEC products have not been tested on the EWSD. Please contact NEC Product Management for any potential sales requiring PRI connection via the EWSD "Custom" interface.*

Note 2: *Support for Operator Access by the AT&T and NTI systems are realized in different ways. The NEAX2000 supports the AT&T implementation.*

Local Exchange Carrier Network Data Services and Features *UNIVERGE* NEAX 2000 IPS Compatibility				
SERVICES AND FEATURES	**LEC CARRIERS**			
Call By Call Service Selection	**AT&T 5ESS**	**NTI DMS100**	**SIEMENS EWSD**	***UNIVERGE* NEAX 2000 IPS**
SWITCHED DATA				
64 kbps Clear Channel	X	X	X	X
64 kbps Restricted	X	X	X	–
56 kbps Restricted	X	X	X	–
HO 384 kbps	–	X	–	–
H11 1526 kbps	–	X	–	–
N x 64 kbps	–	X	–	–
VIRTUAL PRIVATE NETWORK	**AT&T 5ESS**	**NTI DMS100**	**SIEMENS EWSD**	***UNIVERGE* NEAX 2000 IPS**
64 kbps Clear Channel	X	X	–	–
64 kbps Restricted	X	X	–	–
56 kbps Restricted	X	X	–	–
HO 384 kbps	–	X	–	–
H11 1526 kbps	–	X	–	–
N x 64 kbps	–	X	–	–
DEDICATED ACCESS PROVISIONING	**AT&T 5ESS**	**NTI DMS100**	**SIEMENS EWSD**	***UNIVERGE* NEAX 2000 IPS**
SWITCHED DATA				
64 kbps Clear Channel	X	X	X	X
64 kbps Restricted	X	X	X	–
56 kbps Restricted	X	X	X	–
HO 384 kbps	–	X	–	–
H11 1526 kbps	–	X	–	–
N x 64 kbps	–	X	–	–
VIRTUAL PRIVATE NETWORK				
64 kbps Clear Channel	X	X	–	–
64 kbps Restricted	X	X	–	–
56 kbps Restricted	X	X	–	–
HO 384 kbps	–	X	–	–
H11 1526 kbps	–	X	–	–
N x 64 kbps	–	X	–	–

X = Feature Available / – = NOT Available

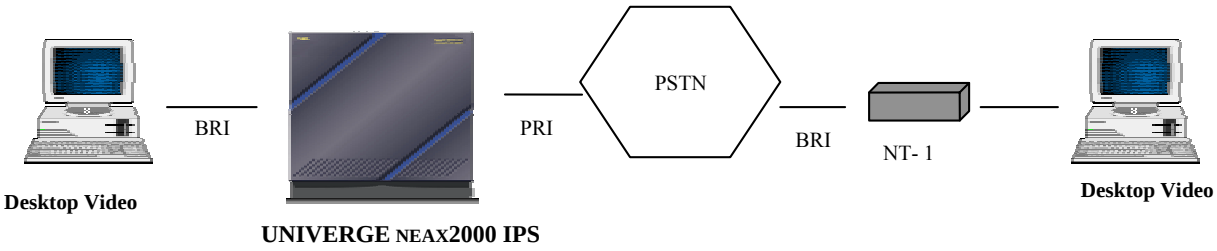
ISDN Basic Rate Interface (BRI)

The *UNIVERGE* NEAX 2000 IPS systems equipped with Basic Rate Interface (BRI) provides the user with a vehicle for circuit switched data at rates of 64 or 128 kbps to the desktop. ISDN BRI connects BRI terminals (stations) to the PBX. ISDN PRI is used to connect the PBX to the public network. Circuit switched data is the focus of the IPS Basic Rate Interface. Features such as Group Four Fax, video conferencing, and data calls are the main objective of BRI on the *UNIVERGE* NEAX 2000 IPS.

Applications

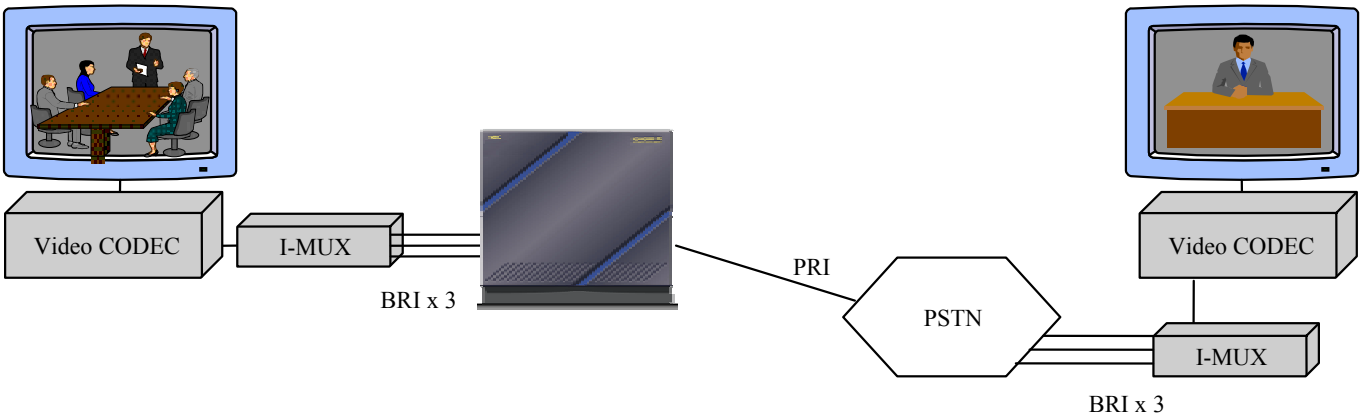
The *UNIVERGE* NEAX 2000 IPS supports National ISDN - 1 and AT&T 5ESS specifications for Basic Rate Interface.

- Desktop Video Applications
  From the desktop, users can place and receive 128 K video conference calls.



- Video Conferencing for large conference rooms or classrooms.

  Using an I-MUX to receive 3 BRI ports, a bandwidth of 384 kbps is possible. This provides good quality video for large conference rooms.



- ISDN Data Modems

  ISDN Data Modems will provide users with access to the Internet at speeds of 64 or 128 kbps. ISDN modems can place and receive ISDN calls at 56, 64, and 128 kbps. Data calls within the PBX or external using ISDN PRI can greatly increase the user's productivity by using the higher rates of ISDN BRI. NEC America has successfully tested the IBM Waverunner series of ISDN modems.

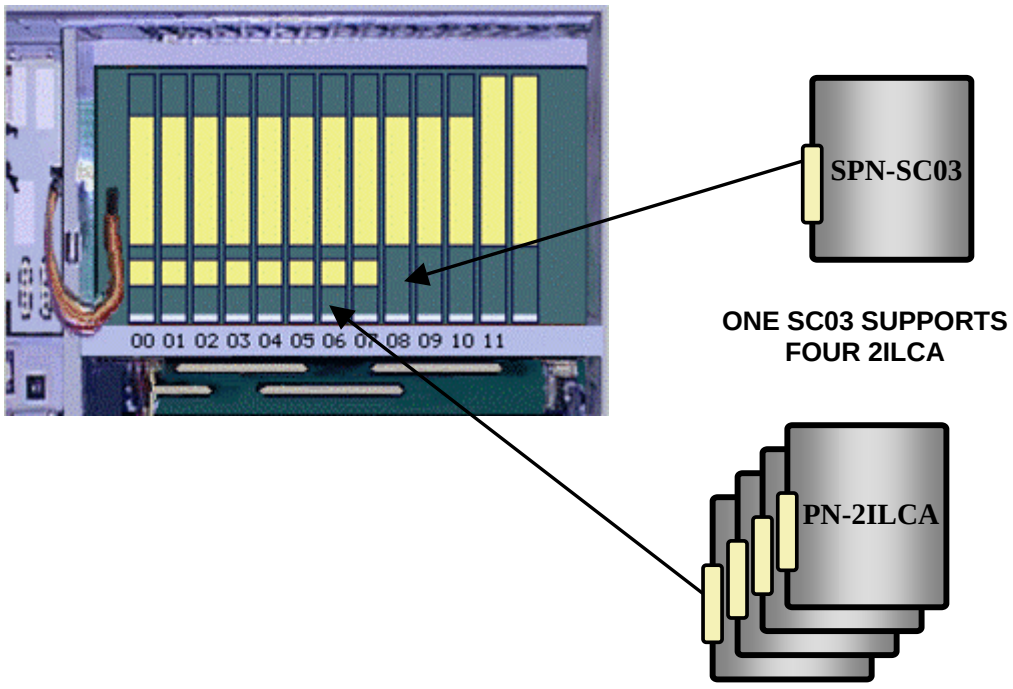
- Group four Fax

  Fax machines capable of G4 can send faxes at speeds of 56 kbps using ISDN BRI.

System Requirements

BRI stations within the IPS can place calls to other BRI stations or outside the PBX by using ISDN PRI lines to the public network. Each IPS system can support up to 24 BRI Terminal ports.

The PN-2ILCA card is used for to connect the BRI terminal to the PBX. An SPN-SC03 card is required to control D channel activity for the BRI terminals. One SC03 card supports four 2ILCA cards. The *UNIVERGE* NEAX 2000 IPS can support 48 ILCA cards that mount in LT slots and 12 SC03 cards that mount in AP slots.



ISDN Equipment List

Card Name	Functional Name	Function
PN-BRTA	BRT	1-line Basic Rate (2B + D) Interface Trunk Card Accommodates one 2-channel PCM digital lines.
PN-4BRTA-F	BRT	4-line Basic Rate (2B + D) Interface Trunk Card accommodates four 2-channel PCM digital lines.
PN-24DTA-C	DTI	Digital Trunk Interface (23B + D, 1.5 Mbps) Card accommodates 24-channel PCM digital lines.
PN-2ILCA	ILC	2-line ISDN Line Circuit Card provides a physical interface to ISDN Terminals. Occupies 8 time slots per one card.
PN-24PRTA	PRT	ISDN Primary Rate (23B + D) Interface Card provides a built-in D-channel Handler (DCH)
PN-SC00	CCH	Common Channel Handler Card transmits/receives signal on the common signaling channel of No. 7 CCIC.
PN-SC01	DCH	D-channel Handler Card transmits/receives signal on the D-channel of ISDN Primary Rate (23B + D) interface or WCS Roaming interface.
PN-SC03	ICH	ISDN-channel Handler Card provides the D-channel signaling interface and controls. Maximum four ILC cards (Layer 2 and 3)

ISDN PRI Required Equipment

Equipment	Description	Qty	Remarks
PN-24DTA-C	24-Channel DTI Card	1-8	
PN-24PRTA	24-Channel PRT Card	1-8	
PN-SC01 (DCH)	D-Channel Handler Card	1-8	1 DTI/Card

ISDN BRI Required Equipment

Equipment	Description	Qty	Remarks
PN-BRTA (BRT)	1-Line BRT Card	1-12	
PN-4BRTA-F (BRT)	4-Line BRT Card	1-24	

ISDN Terminal Required Equipment

Equipment	Description	Qty	Remarks
PN-SC03	ISDN Channel Handler Card	1-24	Supports 4 ILC Cards
PN-2ILCA	ISDN Line Circuit Card	1-48	Supports 2 Terminals

Event Based CCIS Required Equipment

Equipment	Description	Qty	Remarks
PN-SC00	Common Channel Handler Card	1-8	Supports 1 DTI Card

Capacity for ISDN-PRI

Description	Capacity	
	24DTI	24PRT
DTI Card	8	-
DCH Card	8	-
Trunks for DTI	192	-
PRT Card	-	8
Trunks for PRT	-	192
ISDN Routes	8	8
ICH Card	24	24
ILC Card	48	48
Port per DTI Card	24	-
Port per DCH Card	1	-
Port per PRT Card	-	24 + 1 (DCH)
Port per ICH Card	4	4
Port per ILC Card	8	8

System Capacity for ISDN-BRI

Description	Capacity
BRT Card (BRTA/4BRTA-F)	12/24
Trunks for BRT (BRTA/4BRTA-F)	24/192
ICH Card (SC03)	24
ILC Card (2ILCA)	48
Port per BRT Card (BRTA/4BRTA-F)	2/8
Port per ICH Card (SC03)	4
Port per ILC Card (2ILCA)	8

Documentation

More information regarding the ISDN capabilities of the *UNIVERGE* NEAX 2000 IPS can be found in the *UNIVERGE* NEAX 2000 IPS ISDN System Manual and the *UNIVERGE* NEAX 2000 IPS ISDN Features and Specifications.

Chapter 16 Wireless System

UNIVERGE NEAX 2000 IPS Wireless Communication System (WCS)

NEC's wireless technology offers high system capacity, proven system reliability, high voice quality, network integration, investment protection into the future, and extensive user mobility. The *UNIVERGE* NEAX 2000 IPS Wireless Communication System (WCS) provides interface for Personal Station (PS) via a Zone Transceiver (ZT). When installing the WCS, there are three types of Configuration Applications:

1. Integrated Type: Wired for Wireless
   The *UNIVERGE* NEAX 2000 IPS provides both PBX and WCS functions together.
2. Adjunct Type [1]: Analog PBX Interface
   The WCS is positioned as a behind system of the existing PBX linked by LC-COT connection.
3. Adjunct Type [2]: CCIS Interface
   The WCS is positioned as a behind system of the existing PBX linked by CCIS.

Wired for Wireless

This feature allows the *UNIVERGE* NEAX 2000 IPS to have the Wireless PBX feature. The IPS provides several integrated features adding on the adjunctive configuration.

Service Conditions

1. The service conditions of each feature are the same as for the Single Line Telephone, except for the following:

   - Calling Number Display – PS
   - Last Number Redial – PS
   - Speed Dial – PS

2. The station number of a Dterm PS III can be assigned flexibly. The limitation of the number and maximum digits depends on the *UNIVERGE* NEAX 2000 IPS.

Analog PBX Interface

This feature allows the WCS to be integrated with many types of PBXs and to activate various PBX services by sending hook flash signal and DTMF tone.

Service Conditions

1. The signal type of the following cases can be programmed on a route-by-route basis between the WCS and the PBX.

   - To dial for placing a call.
   - To dial for accessing to the PBX service.
   - To dial after sending hook flash.

2. The signal type of dial while a call is established is DTMF. It is not programmable.

3. If the PBX requires pause timing between a trunk access code and an external number, the WCS can automatically insert the pause timing.

4. The called/calling party number is not displayed on the LCD of a Dterm PS III.

5. The interface between a PBX and the WCS is a LC and a COT. The same number of LC and COT as Dterm PS III are required.

6. Code Restriction and Class of Service are provided by the PBX.

7. Any type of call is originated via the PBX, even a Dterm PS III-to-Dterm PS III call.

8. The trunk circuit of the WCS can recognize a momentary open signal as a release signal, if the PBX provides it.

9. The station number of a Dterm PS III can be assigned flexibly. The limitation of the number and maximum digits depends on the *UNIVERGE* NEAX 2000 IPS.

CCIS Interface

This feature allows the WCS to be integrated with NEC PBXs with CCIS interface.

Service Conditions

1. The service conditions of each feature are the same as those of the Single Line Telephone of CCIS features, except the following features:

   - Calling Number Display – PS
   - Last Number Redial – PS
   - Speed Dial – PS

2. If the PBX requires pause timing between a trunk access code and an external number, the WCS can automatically insert the pause timing.

3. The interface between a PBX and the WCS is CCIS. The number of voice link depends on the traffic except PS-to-PS calls.

4. Code Restriction and Class of Service are provided by the PBX.

5. Any type of call is originated via the PBX, even a Dterm PS III-to-Dterm PS III call.

6. The station number of Dterm PS III can be assigned flexibly. The limitation of the number and maximum digits depends on the *UNIVERGE* NEAX 2000 IPS.

7. If the SMDR is needed, AP00 card is required and the upper PBX must have a center node feature of Centralized Billing - CCIS.

Wireless Roaming

The *UNIVERGE* NEAX 2000 IPS supports the JT-Q931a protocol and JT-11582 for signaling at Q-reference point between PBXs on the private network. By supporting this protocol, the PSs can be used in any Calling Area on the private network.

Roaming can be executed only on trunk connection between PBXs based on JT-Q931a protocol or IP. To each trunk route of JT-Q931a digital trunks/IP trunks, it can be specified whether Roaming is provided or not. The JT-Q931a digital trunks/IP trunks can be used by single line telephone stations and Dterm stations for originating or receiving calls as same as common trunks.

The Data Base Module (DBM) card (PN-AP00-B) is required per PBX. The DBM card cannot be used as billing application processor (for SMDR, MCI, PMS or Hotel printer). System data stored in the memory of the DBM card can be saved, loaded and verified from a MAT. A Roaming network can consist of maximum 512 PBXs. Visitor Location Register (VLR) information for maximum 512 Visitor PSs can be recorded to a system. VLR information is the various information of Visitor PS and is made in the memory of DBM on the Visitor PBX when the PS is roaming. When the VLR information exceeds for more than 512 PSs, DBM overwrites the oldest VLR information.

One of the following two Roaming service is available in the PBX.

- Roaming service via D channel private line using Q931a digital trunks; Multi-Site Roaming
- Roaming service via IP network using Virtual IPT; Q931a Roaming over IP Trunk

The following sections describe the system outline and system configuration of Multi-Site Roaming/ Q931a Roaming over IP Trunk.

Multi-Site Roaming

In a mixed network of both *UNIVERGE* NEAX 2400 IPX and *UNIVERGE* NEAX 2000 IPS systems or a network of *UNIVERGE* NEAX 2000 IPS systems supporting wireless users in a private network, some users may desire the ability to take their handsets from one site to the other and still remain in touch with those who are calling. Dterm PS III users can originate or receive a call in any place of a network provided by the plural PBXs which are interfaced by JT-Q931-a, common wireless roaming protocol for network wide roaming capabilities.

Configuration Application: Integrated Type and Adjunct Type with CCIS interface

Service Conditions

1. A roaming service network can be provided by maximum 512 PBXs.
2. One AP00B card for Data Base Module (DBM) is required per PBX.
3. An AP00B DBM-C is required when accommodating 257 or more PS's per site.
4. Visitor Location Register (VLR) information for maximum 512 Visitor PSs can be recorded to a system. When the VLR information exceeds for more than 512 Visitor PSs, AP00 deletes the oldest VLR information in the system.
5. When a call terminates to a Multiline Terminal from a PS, the ringing of the Multiline Terminal is as follows, regardless of Home PS or Visitor PS:

    - When the PS is under control of the same PBX: Ringing for Internal call
    - When the PS is under control of the other PBX: Ringing for External call

6. When the Home PS is roaming, the Home PBX cannot route Announcement-PS No Answer /-PS Out of Zone to the calling party. If the Visitor PBX allowed it will inform the Home PBX that the PS cannot answer, the Home PBX activates Call Forwarding-Not Available.
7. Billing of Home PS is executed to its Virtual LC Number.
8. While the Home PS is roaming, billing information cannot be recorded.
9. The Visitor PS can allow following service features.

    - Originating a call to a station of the same office
    - Originating a call to a station of the different office
    - Terminating a call from a roaming trunk
    - PS Exclusive Hold
    - Call Forwarding

10. The visitor PS cannot allow Calling Name Display.
11. Maximum 64 Visitor PSs can originate or receive a call in a network at the same time.
12. The Visitor PS can answer a call only from the roaming trunk.
13. Billing of Visitor PS is executed to its Virtual LC Number.
14. Billing information for the Visitor PS can be recorded only when the Visitor PS originates a call.

Q. 931a Roaming over IP Trunk

Dterm PS III user can originate or receive a call in any place of a network provided by the plural PBXs, which are interfaced by JT-Q931-a protocol over IP Trunks. Q931a Roaming over IP Trunk is available only on the Point-to-Multipoint connection. Roaming by Q931a digital private line and Roaming by IP network cannot be used together in one PBX. A Roaming IP network consists of maximum 255 PBXs. Roaming via Virtual IPT and Roaming via Physical IPT cannot be used together in one PBX. IP-PAD card is required for the Roaming via Virtual IPT, when the called/calling station is the legacy terminal (Dterm/Single Line telephone/ATTCON, etc.).

Configuration Application: Integrated Type and Adjunct Type with CCIS interface

Service Conditions

1. A roaming service network can be provided by maximum 512 PBXs.
2. One AP00 card for Data Base Module (DBM) is required per PBX.
3. An AP00B DBM-C is required when accommodating 257 or more PS's per site.
4. Both Multi-Site Roaming (using DTI) and this feature cannot be provided in the same system. i.e. the roaming network must be configured in only DTI trunks or IP trunks.
5. Visitor Location Register (VLR) information for maximum 512 Visitor PSs can be recorded to a system. When the VLR information exceeds for more than 512 Visitor PSs, AP00 deletes the oldest VLR information in the system.
6. When a call terminates to a Multiline Terminal from a PS, the ringing of the Multiline Terminal is as follows, regardless of Home PS or Visitor PS:

    - When the PS is under control of the same PBX: Ringing for Internal call
    - When the PS is under control of the other PBX: Ringing for External call

7. When the Home PS is roaming, the Home PBX cannot route Announcement-PS No Answer /-PS Out of Zone to the calling party. If the Visitor PBX allowed it will inform the Home PBX that the PS cannot answer, the Home PBX activates Call Forwarding-Not Available.

8. Billing of Home PS is executed to its Virtual LC Number.
9. While the Home PS is roaming, billing information cannot be recorded.
10. The Visitor PS can allow following service features.

    - Originating a call to a station of the same office
    - Originating a call to a station of the different office
    - Terminating a call from a roaming trunk
    - PS Exclusive Hold
    - Call Forwarding

11. The visitor PS cannot allow Calling Name Display.
12. Maximum 64 Visitor PSs can originate or receive a call in a network at the same time.
13. The Visitor PS can answer a call only from the roaming trunk.
14. Billing of Visitor PS is executed to its Virtual LC Number.
15. Billing information for the Visitor PS can be recorded only when the Visitor PS originates a call.

Wireless Definitions

Virtual Station:	Virtual Station exists only on the system data, provided via non-hardware supported LEN. The Virtual station must be assigned by the system data programming for operating Home PSs and Visitor PSs used for Roaming.
Individual PS number:	Individual PS number is assigned to a PS to identify the PS on the Roaming network. It must be a unique number in the network.
Network ID method:	Network ID method is one method to operate Roaming. A Roaming PS must have two SYS-ID on the Network ID method. One is main SYS-ID for Home PBX, and another is Network ID for Roaming network. The Network ID is used to define whether the PS can operate under the control of PBXs on the Roaming network. The network ID must be the same for all PBXs within the same network.
Visitor PBX:	When a PS leaves control of a PBX to which it belongs originally, and is operating in a zone of another PBX, the PBX is called “Visitor PBX”.
Visitor PS:	When a PS leaves control of a PBX to which it belongs originally, and is operating in a zone of another PBX, the PS is called “Visitor PS”.
Home PBX:	Home PBX is a PBX to which a PS ordinarily belongs.
Home PBX ID:	Home PBX ID is a unique number to identify the PBX on the Roaming network.
Home PS:	When a PS operates under control of a PBX to which the PS originally belongs, the PS is called “Home PS”.
Roaming number:	Roaming number is assigned to a Visitor PS temporarily, when the PS is roaming to a Visitor PBX. The actual Roaming number is Virtual station number assigned as a pilot station of Station Hunting group on the Visitor PBX.
HLR:	Home Location Register. A database to store the location registration data of the Home PS.
VLR:	Visitor Location Register. A database to store the location registration data of the Visitor PS temporarily, when the Visitor PS is in the zone of another PBX.

Wireless – Short Text Message Notification (OAI)

This feature enables a Short Text Message (hereafter, STM), once arrived at a mail box of the STM Server (external equipment), to be automatically distributed to the addressee Dterm PS III via Open Application Interface (OAI). This feature also provides “TM Full” notification on the display (LCD) of the address Dterm PS III that is busy, and if the Dterm PS III is out of zone, the feature makes retransmission of the STM after the PS returns to the zone.

Note: *This describes STM Facilities available for external STM OAI application. Actual functionality depends on the application.*

Configuration Application: Integrated Type

Service Conditions

The feature is available only for My Line (L1) of a two-line Dterm PS III (not provided for Sub Line (L2)).

1. STM transmission route is only one way: STM server fi Dterm PS III
2. One STM can contain 128 byte data at the maximum.
3. The following features are not available at this feature.
   - Call Forwarding - All Calls
   - Call Forwarding - No Answer
   - Call Forwarding - Busy Line
   - Call Forwarding - Not Available

4. If the following occur, STM is not distributed, but stays in the mail box of the STM server. The STM is safely transmitted later by this feature’s retransmission function.

   - Addressee Dterm PS III is out of zone.
   - Addressee Dterm PS III is power off.
   - The line of addressee Dterm PS III is busy, on hold, or ringing.
   - Dterm PS III is in a state of hand-over. **Note**

5. With this feature, PCS Roaming is not available.

Note: *When hand-over unexpectedly starts while the message transmission is partially in progress, this feature sees this as a squelch disconnection and performs message retransmission after the hand-over is completed.*

Features

Feature Matrix by Configuration Type

Feature	Adjunct Type		Integrated Type
	Analog PBX Interface(1)	CCIS Interface(2)	*UNIVERGE* NEAX 2000 IPS Wired for Wireless
Analog PBX Interface	X	–	–
Announcement - PS No Answer	X	X	X
Announcement - PS Out of Cell	X	X	X
Automatic Release - Out of Cell Calls	X	X	X
Call Forwarding - Not Available	–	–	X
CCIS Interface	–	X	–
Calling Name Display - PS	–	X	X
Calling Number Display - PS	–	X	X
DTMF Signal Sender	X	X	X
Group Call Automatic Conference (6/10 Party)	–	–	X
Group Calling - 2 Way Calling	–	–	X
Handover	X	X	X
Individual PS Calling	X	X	X
Last Number Redial - PS	X	X	X
Multi Line Operation - PS	X	X	X
Multi Site Roaming	–	X	X
UNIVERGE NEAX 2000 IPS Wired for Wireless	–	–	X
Number Sharing	–	–	X
Out of Zone Indication	X	X	X
Overlap Dialing	X	X	X
PS Authorization	X	X	X
PS Location Registration	X	X	X
Preset Dialing	X	X	X
Q. 931a Roaming over IP Trunk	–	–	X
Radio Channel Changeover	X	X	X
Short Message Notification (OAI)	–	–	X
Speech Encryption	X	X	X
Speed Dial - PS	X	X	X
Station Hunting - Not Available	–	–	X
Voice Mail Indication	–	–	X

X = available – = not available

Feature Descriptions

Feature	Description
Analog PBX Interface	This feature allows the IPS/WCS to be integrated with many types of PBXs and to activate various PBS services by sending hook flash signal and DTMF tone.
Announcement - PS No Answer /Announcement - PS Out of Zone	This feature allows calls to a Dterm PS III, which cannot be paged, in a predetermined period of time to be routed to the announcement notifying the calling party that the Dterm PS III cannot answer. Announcements can be divided between Announcement - PS No Answer and Announcement - PS Out of Zone depending on the Dterm PS III condition.
Automatic Release - Out of Zone Calls	When a Dterm PS III user engaged in a call moves out of the service area or enters a radio zone where all the channels are busy and the communication becomes unavailable, the call will be disconnected automatically, and the other party receives reorder tone.
Call Forwarding - Not Available	When a PS is power off or out of zone, a call directed to the PS is forwarded to a VMS, and a voice mail message can be recorded to the VMS and checked from the PS. Also the VMS can page the PS automatically after the voice mail message is recorded.
CCIS Interface	This feature allows the IPS/WCS to be integrated with NEC PBXs with CCIS interface.
Calling Name Display – PS	Without answering incoming calls or hold calls terminates to the Dterm PS III, the calling party's name can be confirmed by the indications on the LCD.
Calling Number Display – PS	This feature provides a display on the LCD of a Dterm PS III receiving a call, indicating the station number.
DTMF Signal Sender	This feature allows a Dterm PS III user to send the DTMF signal (tone) to the called party (terminal, voice mail system, etc.) while engaged in communication.
Group Call - Automatic Conference (6/10-Party)	This feature permits a Dterm PS III user, Dterm user or Single Line Telephone user within the system to establish a conference among as many as six or ten parties. From a Dterm PS III/Dterm /Single Line Telephone, a maximum of 9 PSs can be paged simultaneously except the conference leader. The PSs are assigned to the simultaneous paging groups as a participant by the system data beforehand.

Group Call - 2 Way Calling	This feature permits a Dterm PS III/Dterm /Single Line Telephone can page a maximum of fifteen parties simultaneously except the group call leader. After one of paged parties answers, the paging becomes the 2 Way Calling between the leader and the first answered party, and paging other parties stops automatically. The PSs are assigned to the simultaneous paging group as participants by the system data beforehand.
Handover	When the signal transmission quality becomes inferior, a Dterm PS III re-originates a call automatically and seizes another radio channel, making the WCS handover the call to another zone transceiver to maintain the speech quality.
Individual PS Calling	This feature allows the calling party to page the individual Dterm PS III.
Last Number Redial – PS	This feature enables a Dterm PS III to store the numbers dialed in the previous five calls including the last number dialed. The stack dial numbers are sequentially displayed on the LCD, allowing the station user to make an out-going call by selecting the desired dialed number from the display.
Multi-Line Operation – PS	Dterm PS III equipped with two line keys, L1 key and L2 key, and different station numbers can be assigned to each of two lines. The number assigned to L1 key of the PS is called My Line and the other number assigned to L2 key is called Sub Line. My Line and Sub Line of a Dterm PS III can be assigned to appear on the Flexible Line Keys of a Dterm and the Dterm can share the Dterm PS III lines.
Multi-Site Roaming	Dterm PS III users can originate or receive a call in any place of a network provided by the multiple PBXs that are interfaced by JT-Q931a.
***UNIVERGE* NEAX 2000 IPS Wired for Wireless**	This feature allows the *UNIVERGE* NEAX 2000 IPS to have the Wireless PBX feature. *UNIVERGE* NEAX 2000 IPS provides several integrated features adding on the adjunctive configuration.
Number Sharing	This feature allows the Dterm user to have a Dterm PS III as Sub Station and to get service with one telephone number. In case that one user has both Dterm and Dterm PS III, with this feature used, the user is not required to have separate two telephone numbers. ▪ When user is at his desk, a call is terminated to Dterm. ▪ When user leaves his desk with Dterm PS III, a call is automatically terminated to Dterm PS III. In this feature, the Dterm and Dterm PS III are referred as Main Station and Sub Station, respectively.

Out of Zone Indication	When a Dterm PS III user moves out of the service area and the electric field strength becomes weak, this feature notifies it to the user with the warning tone and the LCD display.
Overlap Dialing	This feature allows a Dterm PS III user to receive dial tone and dial the desired number to originate a call.
PS Authorization	This feature is to confirm the identity of a Dterm PS III to avoid an unauthorized Dterm PS III from accessing the system.
PS Location Registration	This feature allows the WCS to supervise the location of each Dterm PS III, upon receiving the location registration request, to allow call termination.
Preset Dialing	This feature allows a Dterm PS III user to confirm the number to be dialed before originating a call.
Q.931a Roaming over IP Trunk	Dterm PS III user can originate or receive a call in any place of a network provided by the plural PBXs, which are interfaced by JT-Q931-a protocol over IP Trunks.
Radio Channel Changeover	This feature is to supervise and changeover the channel automatically to avoid the interference and to maintain the speech quality.
Speech Encryption	This feature protects a call from being tapped.
Speed Dial – PS	This feature allows a Dterm PS III user to dial the certain frequently called numbers using two-digit abbreviated call codes.
Station Hunting - Not Available	This feature allows a call placed to a Dterm PS III station, which is out of zone or power off to be forwarded to an idle station in a hunt group. Two Station Hunting arrangements are available. ▪ Station Hunting - Circular processes the call no matter which station in the hunt group is called. ▪ Station Hunting - Terminal initiates a hunt only when the pilot number of a hunt group is called.
Voice Mail Indication	When a message is mailed in the Dterm PS III, an indication of the envelope icon is displayed in the LCD of the Dterm PS III.

System Description

Dterm PS III

The user terminal for the *UNIVERGE* NEAX 2000 IPS Wireless system is the Dterm PS III. This terminal is specifically designed for the mobile user, offering portability, durability and functionality.

Portability has two aspects to it. First, the handset must be easily carried by the mobile user. At slightly more than 3.8 ounces with dimensions of 5" x 1.5" x 7/8" and with a variety of carrying case options, users can carry the Dterm PS III easily. Second, to be truly portable the handset must also operate over a long period without having to "return to base" for a recharge. With Talk Time at 6 hours and Standby Time at 300 hours, the Dterm PS III will serve through most, if not all, of a typical day on the job.

Durability is another important feature of the Dterm PS III. Once the cord is cut and the user is carrying the handset with them, it will be dropped, banged into cabinets, and generally suffer a fate not experienced by business desktop terminals. Drop-tested to 6 feet, designed to endure hostile environments (temperatures to 122^{o} F), and adaptable to outdoors use with a special case, this handset can meet the needs of most wireless users.

A wireless handset is no good to the user if it does not deliver the features and functions needed for effective communication. The Dterm PS III supports a wide range of timesaving, convenient business features such as:

- Call Hold & Transfer
- Directory Dial-by-name (100 entries)
- Station Speed Dial (10 with name)
- Last Number Redial (20)
- Calling Party Number redial (20)
- Automatic Answer (immediate or delayed)
- Headset Operation
- 4 Line by 18 Digit LCD Display (with scrolling)
- Modem Data (up to 19.2 kbps)
- Vibration Alert

The basic Dterm PS III package includes the Dterm PS III Handset and one Hand Strap. A User Card explaining basic operation and a comprehensive user guide is also available.

Dterm PS III Accessories

To enhance the use of the Dterm PS III, NEC Unified Solutions, Inc. offers several choices in each area of handset accessories. They are:

Lithium-Ion Battery
(PS III Battery Pack) that provides up to 6 hours of continuous talk time and 300 hours on standby. Additional batteries may be required for extended use applications that require "around the clock" coverage.

Battery Charger
Battery Charger for the Dterm PS III includes the Dterm PS III charging slot only. Recharging time is a maximum of 4 hours.

Headsets
Headsets for the Dterm PS III come in two configurations. The first is a classic banded headset with the microphone extension. The second is an ear bud type with an in-line microphone. The banded headset is typically used in cases of prolonged active use (e.g., customer service agent) while the ear set is for occasional use or where concealment is desired (e.g., hotel security).

Dterm Leather Cases
Dterm Leather Cases are available in three models. (1)A Leather Case for carrying the PS III when the large majority of use is indoors. It utilizes a unique integrated swivel clip that allows the handset to rotate out of the way (e.g., in case of sitting), and a quick release mechanism to allow for easy retrieval for use. (2)A Leather Case for carrying the PS III when the large majority of the use is indoors and has an integrated belt clip that doesn't swivel. (3)The Weather Case model offers a weather resistant material for users whose job is outdoors a majority of time.

Zone Transceiver II (ZT)

Access to the communications network by the Dterm PS III user is gained through radio connection with one of the several Zone Transceivers installed on the premise. ZT's are placed strategically around the establishment in an overlapping fashion to provide continuous radio coverage throughout the specified area. The number of Zone Transceivers needed to provide service for any single establishment is defined by several factors:

- The total area requiring wireless service, which is a function of the area of operation of the users of the Dterm PS III handset. Even in the case of providing a small number of handsets, if staff whose job requires wireless service throughout the whole facility uses them, the number of ZT's is determined by the size of the facility, not the number of users.

- The configuration, environment and construction of the building(s) in the areas requiring coverage have considerable impact on the effective coverage area of any one Zone Transceiver. Spacing of ZT's is adjusted according to these conditions resulting in considerably different quantities needed to cover equal amounts of square footage when compared from site to site.

- Special conditions, such as coverage inside elevators and stairwells (both are typically not covered). In addition, accommodating high traffic areas will result in additional Zone Transceivers. Typical deployments provide access to at least 2 ZT's from most locations identified for coverage. By placing ZT's in closer proximity to each other than normal spacing, access to more ZT's in any one spot is supported. In any one zone (an area covered by one ZT, but potentially equipped with more), up to 36 simultaneous calls can be supported within the given 10MHz of the U-PCS band.

- Each Zone Transceiver provides for up to three simultaneous connections to Dterm PS III users within the covered area. As users communicating through a ZT move out of its range and into an area covered by another ZT, the connection is automatically switched (called a handover) to the new ZT (noticeable by a momentary disruption in talk path (**Note**)). With the proper placement of ZT's throughout the facility, users can move about the area while communicating (or idle) and have their calls follow them.

- The ZT is supplied with mounting hardware and can be installed on walls, mounted on solid ceilings or placed above drop ceilings. The Zone Transceiver II is also offered with a weather resistant outdoor enclosure, mountable on outside building walls or poles. Options for both fan cooled or heated enclosures are also available.

- ZT IIU can be deployed using 1 pair or 2 pair wiring.

Note: *With ZT's version 7.2 or higher seamless handover can be achieved.*

Wireless Specifications

Zone Transceiver II Specifications

ITEM	SPECIFICATIONS
• ZT II U	
Output Power	6.8 mW
Type of Interface	U Interface
Number of Simultaneous Speech Path	Maximum 3
Power Supply	• PBX Power Supply (DC –48 V) • Local Power Supply (AC 100 V/200 V)
Power Consumption	Maximum 3 W
Operating Conditions	Ambient Temperature: –10 50 °C Relative Humidity : 45 - 85 % (non-condensing) Installation Location : Indoor
Dimensions (unit: mm)	160 (W) x139 (D) x40 (H) (Excluding Antenna)
Weight	Approx. 400 g

Wire Diameter		26 AWG	24 AWG	22 AWG	19AWG
PBX Power Supply at -48 V	2W	2500 ft. (762 m)	4000 ft. (1219 m)	5500 ft. (1676 m)	8000 ft. (2438 m)
	4W	4300 ft. (1310 m)	6900 ft. (2103 m)	9500 ft. (2895 m)	13000 ft. (3958 m)
PBX Power Supply at -45 V	2W	2000 ft. (609 m)	3200 ft. (975 m)	4700 ft. (1432 m)	7000 ft. (2131 m)
	4W	3500 ft. (1066 m)	5600 ft. (1706 m)	8200 ft. (2499 m)	12200 ft. (3718 m)
PBX Power Supply at -43 V	2W	1800 ft. (548 m)	2900 ft. (883 m)	4400 ft. (1340 m)	6400 ft. (1948 m)
	4W	3100 ft. (944 m)	5000ft. (1524 m)	7600 ft. (2316 m)	11000 ft. (3352 m)
Local Power Supply		11000 ft. (3350 m)	13000 ft. (3958 m)	13000 ft. (3958 m)	13000 ft. (3958 m)

System Capacity

Description		Capacity							
		Integrated			Adjunct (Analog)		Adjunct (CCIS)		
		1PIM	2PIM	3PIM	4PIM	5PIM	6PIM	7PIM	8PIM
Zone Transceiver (ZT)		16	32	48	64	80	96	112	128
Dterm PS **Note**		512							
CSI	4CSIA-A	32							
CSH		24							
PS Simultaneous Connections		216							
Calling Area		32							
ZT per Calling Area		128							

Note: *All existing CSH cards in a system should be SPN-SC03B 8CSH-C (AP) when accommodating 257 or more PSs.*

WCS Required Equipment

EQUIPMENT	DESCRIPTION	QTY	REMARKS
PZ-PW122 (DC/DC PWR)	–48V Power supply card for ZT One per PIM 16 ZT powered/card	1-8	Analog Interface CCIS Interface Wired for Wireless
PN-4CSIA-A (CSI)	ZT Interface card 4 per PIM	1-32	
PN-SC03-B (CSH)	ZT Handler card 1 per 2 4CSIA cards	1-16	
PN-4COT PN-8COT	COT for Analog Interface 4COT: 4 PS per card 8COT: 8 PS per card	4COT: 64 8COT:32	
SPN-24CCTA	1.5M AMI (24ch) CCIS Interface Trunk	As req. Max. 8 per sys	
PN-24DTA-C(24DTI)	24-channel DTI card	1-8	For Multi-Site Roaming **Note**
PN-30DTC-C (30DTI)	30-channel DTI card	1-8	
PN-SC01 (DCH)	D-channel Handler card	1-8	
PN-AP00-B (DBM)	Roaming Data Base Module card	1	
PZ-M606-A (ETHER)	Ethernet Control card	1	For Q931a Roaming over IP Trunk (Virtual IPT)
PN-32IPLA (IP-PAD)	32-channel IP-PAD card	1-8	
PN-16VCTA (16VCT)	16-channel CODEC card	1-16	
IPTRK BUS CA	Set of 3 BUS Cables between IP-PAD and 16VCT cards Length: 0.1 m (0.33 ft.)	1-8 sets	
PN-8IPLA (IP-PAD)	8-channel IP-PAD card	1-8	
PN-24IPLA (24DSP)	24-channel expansion DSP card for 8-channel IP-PAD card	1-8	

Note: *The SPN-AP00B DBM-C is required when accommodating 257 or more PSs.*

WCS Required Equipment (Cont.)

EQUIPMENT	DESCRIPTION	QTY	REMARKS
• 10BASE-T Cable (TIA/EIA category 3 or larger) • 100BASE-TX Cable (TIA/EIA category 5)	10BASE-T/100BASE-TX Cable between ETHER card and Ethernet Cable Length: Maximum 100 m (328 ft.)	As required	For Q931a Roaming over IP Trunk (Virtual IPT) **Note**
Router		As required	
PN-AP00-B (DBM)	Roaming Data Base Module card	1	
PN-IPTB (IPT)	IP Trunk card	1-8	For Q931a Roaming over IP Trunk (Physical IPT)**Note**
PN-4VCTI (4VCT)	4-channel CODEC card	1-32	
IPTRK BUS CA	Set of 3 BUS Cables between IPT and 4VCT cards Length: 0.1 m (0.33 ft.)	1-32 sets	
• 10BASE-T Cable (TIA/EIA category 3 or larger) • 100BASE-TX Cable (TIA/EIA category 5)	10BASE-T/100BASE-TX Cable between IPT card and Ethernet Cable Length: Maximum 100 m (328 ft.)	As required	
Router		As required	
PN-AP00-B (DBM)	Roaming Data Base Module card	1	
PZ-M606-A (ETHER)	Ethernet Control Card	1	For Short Message Notification (OAI)
• 10BASE-T Cable (TIA/EIA category 3 or larger) • 100BASE-TX Cable (TIA/EIA category 5)	10BASE-T/100BASE-TX Cable between ETHER card and Ethernet Cable length: Maximum 100 m (328 ft.)	1	

Note: *The SPN-AP00B DBM-C is required when accommodating 257 or more PSs.*

Wireless LAN

The current Wireless LAN supported interface to the *UNIVERGE* NEAX 2000 IPS is the *UNIVERGE* WL System Telephony Gateway. The *UNIVERGE* WL System Telephony Gateway is a wireless telephony product that provides high quality packetized voice communications using Internet Protocol (IP). This configuration uses a Digital Interface from the *UNIVERGE* NEAX 2000 IPS to integrate with the mobile handsets to emulate digital telephone sets to deliver advanced capabilities such as multiple line appearances and LCD display features.

Wireless LAN Handset

The MH110 and MH120 Wireless Telephones are mobile handsets for workplace telephone systems. The Wireless Telephones operates an 802.11b (802.11 for FH) wireless Ethernet LAN providing users a wireless voice over IP (VoIP) extension. By seamlessly integrating with the Telephone Gateway and the site PBX, Wireless Telephone users are provided with high-quality mobile voce communication throughout the workplace. The Wireless Telephones gives users the freedom to roam throughout the workplace while providing all the features and functionality of a desk phone.

The Wireless Telephones reside on the wireless LAN with other wireless devices using Direct Sequence (DSSS) or Frequency Hoping (FHSS) Spread Spectrum radio technology. The handset radio transmits and receives packets at up to 11Mb/s.

The Wireless Telephone supports Wired Equivalent Privacy (WEP) as defined by the 802.11 specification. NEC Unified Solutions, Inc. offers the product with both 40-bit and 128-bit encryption. WEP increases the security of the wireless LAN to the level similar to a wired Ethernet LAN.

Wireless LAN Documentation

For details on Wireless LAN related hardware and configurations refer to system documentation online at the following links. Some or all links may require a valid login and password for access.

http://www.ilibrary.com/cng/docs.nsf (See System Documentation)

http://www.necunifiedsolutions.com/main/Products/ProProductCategoryMain.asp?catid=161

Chapter 17 Hotel/Motel System

Hospitality features are included in the basic system software of the *UNIVERGE* NEAX 2000 IPS. Additional software is not required. Any Business system can use the Hospitality features and any Hospitality system can use any Business features as necessary.

The *UNIVERGE* NEAX 2000 IPS Hotel/Motel features allow the hotel's administrative staff to control the telephone activity and check status of guest rooms. Service staff can also communicate the condition of the guest rooms back to the front desk. By dialing access codes from the guest room stations, the service staff can report to the front desk area that a room Needs repair, Repairs are complete, Cleaning in progress, Cleaning complete, Room has been checked, and other important service functions. By dialing access codes from the guest rooms, the time spent dialing, waiting for answer, and communicating the message to someone is eliminated.

The *UNIVERGE* NEAX 2000 IPS Hospitality package offers the features that will be described in detail in the following sections.

Hotel Feature List

Feature Name	Application				
	Guest	Administrative	Front Desk Terminal	Hotel Console	PMS
Automatic Wake Up	X	X	X	X	X
Check In/Check Out **Note**	–	–	X	–	X
Direct Data Entry	X	–	–	–	X
Do Not Disturb-Hotel/Motel	X	X	X	X	X
Do Not Disturb-System	–	–	X	X	–
Hotel/Motel Attendant Console	–	–	–	X	–
Hotel/Motel Front Desk Instrument	–	–	X	–	–
House Phone	X	X	X	X	–
Maid Status	X	X	X	–	X
Message Registration	–	–	X	–	X
Message Waiting	–	X	X	X	X
Property Management System (PMS) Interface	–	–	–	–	X
Room Cutoff	–	–	X	X	X
Room Status	–	–	X	X	–
Single Digit Dialing	X	X	X	X	–

X: Applicable –: Not applicable

Note: *When MP built-in PMS on IP is provided, you can set and cancel this hotel feature only from PMS.*

Application Processor

- PMS with AP00 on RS-232C
  The Application Processor (AP) manages guest or administration room status and stores call information on each guest or administrative station. The AP also provides interface ports for a Property Management System (PMS), Station Message Detail Recording (SMDR) and a Hotel Printer.

- MP built-in PMS over IP
  The Main Processor (MP) manages guest or administration room status and stores call information on each guest or administrative station. The MP also provides a LAN interface port for a Property Management System (PMS) terminal or a Station Message Detail Recording (SMDR) terminal. Adding an AP00B MRC-F with PMS over IP allows the print out of an execution history of Automatic Wake Up, Do not Disturb, Room Cutoff, Check-in/Check-out service, Message Waiting and PMS connect or disconnect status.

Hotel Console

The DESKCON is programmed as a Hotel Console. The Hotel Console can access Room Cutoff (individual and group), Automatic Wake Up, Message Waiting, or Do Not Disturb (individual and group) with the function keys; in addition to the attendant features and functions.

Hotel/Motel Front Desk Terminal

The Dterm can be programmed to function as a Hotel/Motel Front Desk Terminal. This allows setting and canceling of the following hotel features:

- Automatic Wake Up
- Check In/Check Out **Note**
- Do Not Disturb
- Do Not Disturb-Override
- Message Waiting
- Room Cutoff
- Room Status **Note**

Note: *When MP built-in PMS on IP is provided, you can set and cancel these hotel features only from PMS.*

Station Message Detail Recording (SMDR)

The Station Message Detail Recording (SMDR) sends out the outgoing/incoming C.O. call information to an external SMDR terminal (Personal Computer). The SMDR is usually used in conjunction with the PMS and used for the following purposes.

- Management of guest/administrative station call
  - The PMS does not manage the guest/administrative station call
- Backup of guest/administrative station call for a PMS failure
- Management of either guest or administrative station call
  - For example, the SMDR manages an administrative station call, and the PMS manages a guest station call

Hotel Printer

When PMS with AP00 on RS-232C is provided, the various system messages and the guest room status can be obtained through a locally provided Hotel Printer. The following information is automatically printed out as a system message:

- Wake Up attempts whether successful or not
- Remaining messages for the station which are set to Check Out
- Codes and quantities of the goods requested from a guest room by Direct Data Entry

If the print out function key is provided on the Front Desk Terminal, the status of the following features are printed out when the feature is set or reset and Room Status print out is activated:

- Automatic Wake Up
- Check In/Check Out
- Do Not Disturb
- Message Waiting
- Room Cutoff
- Room Status-individual guest station/all guest stations

Executed Task Print Out

Using the built in PMS on the MP and an AP00B MRC-F will allow the print out of an execution history for Automatic Wake Up, Do not Disturb, Room Cutoff, Check-in/Check-out service, Message Waiting and PMS connect or disconnect status.
In the following cases, a history can be output:

- When Automatic Wake Up is set/cancelled
- When Automatic Wake Up from an administrative station or console is set/canceled
- When Automatic Wake Up has been executed
- When Do Not Disturb is set/cancelled
- When Room Cutoff is set/canceled
- When Message Waiting is set/canceled
- When a check-in/checkout/checkout cancellation is performed from PMS
- When PBX-PMS communication is started/disconnected

Features

Automatic Wake-up

This feature allows the system to be programmed to automatically call guest rooms or administration stations at specified times. Upon answering, the guest is connected to a recorded announcement or music source. A printout of Unanswered, Busy or Blocked Automatic Wake-Up attempts for each guest room is provided using the Hotel/Motel printer.

Setting and canceling can be initiated from the following:

- Attendant Console
- Hotel/Motel Front Desk Instrument
- Guest Station
- Administrative Station
- Property Management System (PMS) terminal

1. The time is entered on a 24-hour system in one-minute increments.

2. A maximum of 32 stations can be set for the same Automatic Wake-Up time. When the setting exceeds 32, the excess stations will be set to an earlier time in five-minute intervals.

3. Wake-Up attempts, whether successful or not, can be printed out at a locally provided printer. The results of execution of Automatic Wake-Up set and cancel are also printed.

4. The Automatic Wake-Up call will ring a station in Do Not Disturb.

5. When Automatic Wake-Up call is received, the station receives music or an announcement. A DAT or COT card (with locally provided music or announcement source) is required. As an option, a PN-DK00 card can be programmed to provide a contact closure for starting the external announcement or music source when used with a COT card.

6. Data for Automatic Wake-Up programming is canceled by Check Out operation.

7. The number of Automatic Wake-Up call attempts is programmable from one to five times.

8. If the station does not answer, is busy, in Line Lockout, or ringing, recalling is initiated one minute later. Recalling is repeated up to five times. Each call that fails is printed. When the final attempt results in failure, a buzzer is sounded at the printer and a flashing LED on the DSS/BLF Console is available to indicate which station does not answer.

9. Call Forwarding and other features will not reroute Automatic Wake-Up calls.

Check In / Check Out

When this feature is activated, the following operations occur:

Check In:
Room Cutoff is cleared.

Check Out:
Room Status printout is supplied.
Do Not Disturb is reset.
Room Cutoff is set.
Message Waiting is reset.
Automatic Wake Up is cleared.

Multiline Terminals with LCD can be assigned as Hotel/Motel Front Desk Instruments. This feature can only be activated from the Hotel/Motel Front Desk Instrument or the Property Management System (PMS). This feature only applies to guest room stations. Refer to the Hotel/Motel Front Desk Instrument and Property Management System Features and Specifications for more information on Check In/Check Out.
Note: *When MP built-in PMS on IP is provided, you can set and cancel these hotel features only from PMS.*

Direct Data Entry

This feature allows a maid or other hotel personnel to enter numeric data to the Property Management System (PMS), using the guest room station for entry through dial operation. The same numerical data can be output to a Hotel/Motel Printer by system data programming.

This feature is activated from a guest room telephone.
The digits 0, 1, 2, 3, 4, 5, 6, 7, 8, and 9 can be used for the input data.

The input data can be sent out to the PMS and/or Hotel/Motel Printer. The output option can be selected by system data programming.

A maximum of 30 digits can be entered in one operation to the PMS and/or Hotel/Motel Printer. If data exceeds 30 digits, the guest room station receives reorder tone.

If the input data to be sent is less than 30 digits, "#" can be used to end the digit string. (The "#" is not sent to the PMS and Hotel/Motel Printer.)
Example: Access code + 1234567890#

When the Hotel/Motel Printer is used, "*" can be used to separate the data entry ("*" is not printed out, but is sent out to the PMS). In this case, the input data including "*" is up to 30 digits.
Example: Access code + 001*002*12345*3#

Reorder tone will be received instead of service set tone if the PMS does not respond within 15 seconds or sends a negative answer to the *UNIVERGE* NEAX 2000 IPS.

Do Not Disturb Console

This feature allows a DSS/BLF Console associated with a Multiline Terminal to be used as a Do Not Disturb (DND) Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. DND set status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/cancel the DND status of other stations using the DND Console.

Do Not Disturb-Hotel/Motel

This feature allows the Attendant Consoles, Hotel/Motel Front Desk Instruments, guest stations or Property Management System (PMS) terminals to place individual stations into Do Not Disturb. Calls can be placed from stations set in DND.

- Automatic Wake Up and Timed Reminder will override Do Not Disturb.
- A station in Do Not Disturb can be called from the Attendant Console or the Hotel/Motel Front Desk Instrument using the DNDOVR key.
- Do Not Disturb is automatically cleared when Check Out is performed.
- Depending on system programming, an incoming call addressed to a station in DND condition is routed to one of the following on a per tenant basis:
  - Reorder tone
  - Attendant Console
  - A pre-assigned station

- Call Forwarding - Busy settings by stations in DND will result in calls being forwarded, even if the stations are idle.

Do Not Disturb-System

This feature simultaneously restricts incoming calls to a pre-assigned group of stations by operation from the Hotel/Motel Front Desk Instruments. Attendant Consoles and Hotel/Motel Front Desk Instruments can use the DND OVR key to override this Do Not Disturb setting.

- Stations are assigned to the Do Not Disturb (DND) Group in Class of Service.
- Calls to extensions whose stations are in Do Not Disturb will receive reorder tone.
- The station in Do Not Disturb can originate calls in the normal manner.
- Verification of stations in Do Not Disturb is possible from the Hotel/Motel Front Desk Instrument and Attendant Consoles.
- Only Hotel/Motel Front Desk Instruments and Attendant Consoles have the ability to place a group of stations in Do Not Disturb. There is only one group available, and the stations within the group are programmed in system data. There is no limitation on the number of stations in the group.
- A station included in a DND group retains the ability to place that particular station in DND.
- When the Hotel/Motel Front Desk Instrument places a group in DND, an individual station within the group can cancel the DND setting to that station.

Hotel/Motel Attendant Console

The Attendant Console can be programmed to function as a Hotel/Motel Attendant Console. In addition to the business features and functions of the Attendant, the Hotel/Motel Attendant Console can set Room Cutoff (individual and group), Automatic Wake Up, Message Waiting, and Do Not Disturb (individual and group).

Hotel/Motel Front Desk Instrument

A Multiline Terminal with LCD can be programmed to function as a Hotel/Motel (H/M) Front Desk Instrument. This can be used to set and cancel standard H/M features such as the following:

- Do Not Disturb
- Room Cutoff
- Message Waiting
- Print Out
- Check In/Check Out
- Automatic Wake up
- Room Status
- Do Not Disturb Override

For each feature desired, a dedicated line key on the Hotel/Motel Front Desk Instrument must be assigned.

The Room Cutoff and Do Not Disturb features are applicable to a group of stations and individual stations. There is one group available for each feature and the stations in the groups are programmed on a per-station basis in system data.

The Print Out function provides a hard copy on the status of the following features when the feature is set or reset:

- Do Not Disturb
- Room Cutoff
- Message Waiting
- Automatic Wake up
- Room Status
- Check In/Check Out

Successful and unsuccessful Wake Up attempts are printed out.

The Print Out function allows selection of output based on individual station numbers (except for Room Status).

When Check Out is done, the following functions are set or cleared:

- DND – cleared
- Message Wait – cleared
- Room Cutoff – set
- Automatic Wake Up - cleared

Room Status Codes are totally flexible, and the user determines the meaning for each code. The system will print the maid ID or station number (if other than guest room) that set the specific code. Up to eight codes (1-8) are available.

House Phone

This feature allows selected stations to reach the Attendant simply by going off-hook.

- House Phones may be equipped without dials.
- House Phone assignments are programmed into system data from the Maintenance Administration Terminal (MAT) or the Customer Administration Terminal (CAT).
- There is no limit to the number of House Phones permitted in the system provided the maximum amount of available ports is not exceeded.
- The Attendant has the option to process the call using the delay or non-delay operation or passing dial tone to House Phones with dials.
- The system's response to a hook flash from a House Phone can be assigned to provide feature dial tone or recall the Attendant.
- Four house phone groups are available.
- Each house phone group can be assigned to automatically call the Attendant Console or a predestinated station.

Maid Status

This feature allows the Hotel/Motel (H/M) Front Desk Instrument, Property Management System (PMS) terminal, or guest room station (using special access code) to register the condition of each guest room. There are a maximum of eight types of room conditions.

Although the system provides for a maximum of eight types of room conditions, the eight codes (1-8) provided are not fixed for a given condition. This provides the end user with the flexibility to select any code (1-8) to serve as the Maid Status code for any condition. When Maid Status is displayed, or printed out, the digit(s) dialed as the code when the Maid Status was last changed are displayed or printed.

When dialing from a guest station, a Maid Identification Code (maid ID code) can be provided (up to two digits). This is allowed or denied in system programming. Status number Room condition (example; see below)

1. Check In
2. Check Out
3. Under Cleaning
4. Cleaning Finished
5. Check Finished
6. Out of Service
7. Needs Maintenance
8. Maintenance Complete

Message Registration

This feature provides output from the IPS to the Property Management System (PMS). This allows the PMS system to receive call record reports for guest room phone calls. The PMS system can combine the phone charges with the room charges, allowing for a presentable final bill to the guest upon check out.

Message registration is a condensed SMDR record, designed to provide only the necessary call information to the PMS system.

This feature is a function of the PMS feature and is optional. The standard SMDR output can be used along with this feature.

Message Waiting

This feature allows the Attendant Console, Hotel/Motel (H/M) Front Desk Instrument, administrative station, or Property Management System (PMS) terminal to light a lamp (on an uninterrupted or interrupted basis) on a Single Line Telephone or Multiline Terminal to indicate a message is waiting.

In addition to the lamp indication control, this feature also provides the Voice Message Waiting service that an originating station user can set the Message Waiting with a prerecorded message by using the Digital Announcement Trunk card (PN-2DATA)

Voice Message Waiting - System

An originating station user can choose the prerecorded message to be set by dialing the message number associated. The predetermined station records the messages.

Voice Message Waiting - Individual

When setting Message Waiting, an originating station user announces the message to be recorded after dialing the station number.

Message Waiting Console

This feature allows a DSS/BLF Console associated with a Multiline Terminal to be used as a Message Waiting (MW) Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. The Message Waiting status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/reset MW status using the MW Console.

The MW Console only displays/cancels the set status for those stations to whom Message Wait was set, by the station user of the MW Console. If the system is initialized (reset), the Console function mode will return to DSS/BLF.

Property Management System (PMS)

The *UNIVERGE* NEAX 2000 IPS provides a data interface to a customer-supplied Property Management System (PMS) accommodating Hotel Management Features. The PMS (Model 60 only) can be any computer connected to the PBX via a RS-232C Interface. It communicates with the PBX using the specified protocols.

The Data Link Interface allows the PMS to accommodate both front and back-office Hotel Management Features, by providing a means of communication between the PMS and the PBX for features such as Check In/Check Out, Message Waiting, Station Message Detail Data, and control features such as Do Not Disturb and Room Cutoff.

The PMS can communicate with the PBX to obtain the following information:

1. Maid Status:
   This information can be entered from either a guest room telephone or Front Desk Instrument, and will automatically be transmitted to the PMS for data update.
2. Message Waiting Lamp Status Change
   This information can be entered from the Attendant Console or Front Desk Instrument. It is then automatically transmitted to the PMS for data update. If the automatic MW lamp off feature is activated, MW data is cleared and status is sent to PMS.
3. Station Message Detail Data
   This information is transmitted to the PMS after completion of each local and toll call.
4. Wake-Up Service
   This information can be entered from the Attendant Console, Front Desk Instrument or guest room station, and is automatically transmitted to the PMS for data update.
5. Do Not Disturb/Room Cutoff
   This information can be entered from the Attendant Console or Front Desk Instrument, and is transmitted to the PMS by request from the PMS.
6. Check-In/Check-Out
   When PMS with AP00 on RS-232C is provided, this information can be entered from the Attendant Console or Front Desk Instrument, and is automatically transmitted to the PMS for status update.
7. Room data image messages indicating requests for database updates and database images.
8. Room change, room swap and room copy for data update. **Note**
9. Room occupancy change and room data change for data update.
10. Routine activity checks between the PMS and the PBX.

Note: *Room copy is available only when MP built-in PMS over IP is provided.*

The PMS can send the following information to the PBX:

1. Maid Status
2. Message Waiting Lamp Status Changes
3. Telephone Restriction Status Changes
4. Check-In/Out Messages
5. Room Data Image Inquiry
6. Wake-Up Status Changes
7. Room Change, Room Swap and Room Copy **Note**
8. Room Occupancy and Room Data Change
9. Status inquiry for routine activity checks
10. Guest Name and Guest Information to be displayed on Administration Station, Front Desk Instrument or Attendant Console **Note**

Note: *Room copy and Guest Room Information is available only when MP built-in PMS over IP is provided.*

Room Cutoff

This feature allows the Attendant Console, Hotel/Motel (H/M) Front Desk Instrument, or Property Management System (PMS) terminal, or guest room telephones using a special access code, to temporarily restrict guest room telephones from making unauthorized calls when guests are away from their room, and when rooms are in Check Out status. There are two types of Room Cutoff conditions depending on the type of calls restricted.

- External Call Restriction: All outgoing calls from guest room stations are restricted in the Room Cutoff status.
- Toll Call Restriction: All toll calls from guest room stations are restricted during Room Cutoff status. (Internal and local calls are available.)

Stations in Room Cutoff condition are able to place outgoing calls using the Attendant Assisted Calling feature. If the station under Room Cutoff status dials a C.O. line access code and/or a special area code, the station is rerouted to one of the following:

- Reorder tone
- Attendant Console

Room Cutoff is automatically set by Check Out operation and it is automatically reset by Check In operation. Station-to-station calling and service feature access (such as Maid Status) are still available.

Room Cutoff Console

This feature allows a DSS/BLF Console associated with a Multiline Terminal to be used as a Room Cutoff Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. The Room Cutoff status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/cancel Room Cutoff to another station using the Room Cutoff Console.

The Room Cutoff Console only displays/cancels the set status for those stations that were set into room cutoff by the station user of the Room Cutoff Console. Indication is not provided when another Room Cutoff console, the Attendant Console, a Hotel/Motel Front Desk Instrument, or a PMS changes the RC status of a station.

Room Status

This feature provides the Hotel/Motel (H/M) Front Desk Instrument with a visual display of the guest's room status. A supplementary print out (individual and summary) can be provided.

Items indicated are as follows:

- Set status of Do Not Disturb
- Set status of Message Waiting
- Set status of Room Cutoff
- Automatic Wake Up Time if set
- Check In/Check Out status
- Maid Status

The status of the function is indicated by a green LED associated with each function key:

- Check In/Check Out set
- Do Not Disturb set
- Message Waiting set
- Room Cutoff set
- Automatic Wake Up set (LCD displays time set)

The Room Status of stations can be printed at the Hotel/Motel printer (if available) by pressing the print (PR) key on the Hotel/Motel Front Desk Instrument prior to pressing the STS key.

Single Digit Dialing

This feature provides the station user the ability to dial single digit codes to access certain features while still allowing the same digit dialed to be used as the first digit of guest room station numbers.

This feature is available on a numbering plan basis. Up to four different numbering plans are available per system. For multiple tenant applications, each tenant can be assigned to one of these four numbering plans.

When this feature is assigned, digits in the numbering plan can overlap and Single Digit Dialing is based on a time-out after dialing the first digit. The timing duration before the system stops looking for a second digit is programmable from two-eight seconds. The default setting is four-five seconds.

Digits 0-9, *, and # can be assigned within each numbering plan for Single Digit Dialing.

The following features can be activated using Single Digit Dialing:

- Trunk Answer Any Station
- Trunk Access
- Single Digit Station Numbering
- Operator Calls (Dial 0)

Hotel System Capacity

Description	Capacity
Guest/Administrative Station	1020
Front Desk Terminal	8
Hotel Console	8
Hotel Printer	2
I/O port for PMS/SMDR/Hotel Printer	2
I/O port for SMDR/PMS via LAN	1
I/O port for Hotel Printer	2

System Specifications

- PMS/SMDR via RS-232C Interface (Model 60 only) and Hotel Printer Interface

Item	Specifications	
	PMS/SMDR Interface	Hotel Printer Interface
Physical Interface	RS-232C	RS-232C
Synchronization	Asynchronous	Asynchronous
Protocol	IMS Procedure	—
Transmission Speed	1200/2400/4800/9600 bps **Note**	1200/2400/4800 bps
I/O port	No. 0-3 port of AP00-B card	No. 0/3 port of AP00-B card

Note: *For the port 1 and port 3 of AP00-B card with AP00 program, data speed cannot be set to 9600 bps.*

- PMS/SMDR via LAN Interface (Model 90/120 only)

Item	Specifications
Physical layer	Ethernet
Connection layer	Ethernet packet format complies with the DIX standard.
TCP/IP protocol	ARP, IP, ICMP, UDP, TCP
Socket interface	Complies with 4.3 BSD socket interface
Transport protocol	TCP stream type protocol
Application port number	SMDR: 60010 (fixed) PMS : 60050 (fixed)
Number of connections	1
Client/Server	Client : SMDR/PMS terminal Server: PBX
Transmission code	7-bit ASCII code
Quasi-normal restriction condition	1. When connection is closed 2. Status monitoring text

Station Equipment

Front Desk Instrument:	Dterm Series E/Series i with Display Terminals with 16 feature keys are recommended
Guest Room Phone:	Analog Single Line H/M Analog Single Line Dterm Series E Digital 2-Line
Hotel/Motel Attendant Console:	SN716 DESKCON
House Phone:	Analog Single Line Dterm Series E/Series i Digital Terminal
Administration Phone:	Analog Single Lines Dterm Series E/Series i Digital Terminal

Chapter 18 Call Accounting

AIM*WorX*™

NEC's Applications Integration Manager or AIM*WorX* is an application suite of management software based on a call accounting foundation. It provides an easy way to cost calls with a variety of features to use this information. AIM*WorX* has a wide range of predefined administrative reports and gives you the ability to create your own.

AIM*WorX* first gathers Station Message Detail Records (SMDR) from the switch. This data stream is sent each time a call is completed. It contains data about the call including; trunk and route information, phone numbers of where the call was placed from and where it went to, what time it started and ended, and variable information that may include authorization codes, account codes, or office codes. AIM*WorX* costs each record using costing data that you have defined to create Call Detail Records (CDR). These records provide the data to run queries and reports.

AIM*WorX* supports many Single Point of Entry (SPE) plug-in modules and add-on applications. Single Point of Entry modules share AIM*WorX* databases, allowing user information to be entered once for many different applications. These modules add interfacing capabilities to remote functions through tabs added to the AIM*WorX* screens.

Add-on applications are telecommunication packages that have separate user interfaces but can use AIM*WorX* already established database. Examples of add-ons are Interactive Directory, Message Center, and Visual Console.

All optional features, plug-ins, and applications are delivered on a single CD ROM. Included, as part of the package is a security mechanism (hardware key) that allows the customer to register the product, and to activate the desired features for the product that they selected. Also contained on the CD ROM are online help files for each application, and a copy of all associated application documentation that can be read by the customer using a shareware tool included for this specific purpose.

AIM*WorX* is available as software only for use with customer-provided PC, or as a turnkey hardware system. These turnkey systems are pre-loaded at our customizing center on servers and desktop PCs from NEC Computer Systems Division (NEC CSD).

AIMWorX integrates with Property Management Systems when the AIMWorX Hospitality Links module is purchased.

Multi-Site (Centralized Operations)

Multi-Site operation allows AIM*WorX* Call Accounting to collect data from more than one SMDR source. Simply put, a site is a PBX or group of PBXs. If you have several separate PBXs (operating independently, whether on- or off-premises), Multi-Site operation is required to let AIM*WorX* collect the call data and process it for each site. Multi-Site collects SMDR via polling, direct connection, or a combination of both. How AIM*WorX* handles each PBX is configured in its PBX setup screens.

AIMWorX databases

AIMWorX supports both Microsoft SQL databases. You may purchase client/server version of AIM*WorX* with SQL Server 2000 database or without a database. AIM*WorX* demonstration systems may use the free version of SQL Server called Microsoft Database Engine (MSDE). This is included on the AIM*WorX* CD.

Select Database Licensing Option	
Description	**Notes**
AIMWorX Standalone DB	AIMWorX Program Disk with Microsoft SQL 2000 Integrated Disk 1 Client Access License
AIMWorX C/S NEC Provided DB	AIMWorX Program Disk with Microsoft SQL 2000 Integrated Disk 5 Client Access Licenses
AIMWorX C/S Cust. Provided DB	AIMWorX Program Disk No Database - Customer provided

AIMWorX Standalone DB: The AIM*WorX* Standalone Database includes the AIM*WorX* Program disk, which has all modules on it. The core modules that are enabled are AIMWorX Manager (Call Accounting) and Alarm Manager (included, is the freely licensable Alarm Client). The standalone version only allows one person to access the AIM*WorX* application at a time, from the AIM*WorX* server.

AIMWorX C/S NEC Provided DB: The AIM*WorX* Client/Server NEC Provided Database includes the AIMWorX Program disk, which has all modules on it. The core modules that are enabled are AIM*WorX* Manager (Call Accounting) and Alarm Manager (included, is the freely licensable Alarm Client). Processor licensing allows an unlimited number of clients to access the AIMWorX applications simultaneously from remote PCs connected to the LAN. Access from PCs over the Internet is also possible, if the proper TCP/IP port is enabled through an organizations firewall. Refer to AIM*WorX* documentation for more information on this.

AIMWorX C/S Customer Provided DB: The AIM*WorX* Client/Server Customer Provided Database includes the AIM*WorX* Program disk, which has all modules on it. The core modules that are enabled are AIM*WorX* Manager (Call Accounting) and Alarm Manager (included, is the freely licensable Alarm Client).

SQL Server licensing

If you purchase SQL Server with AIMWorX, the licensing is provided in multiples of five Client Access Licenses (CALs). All AIMWorX Client/ Server versions include the first five CALs. Purchase additional CALs in multiples of five.

You may also choose to license the SQL Server database in processor licenses. You purchase one processor license for each processor in the AIMWorX server and receive unlimited user login name assignments. Purchase additional processor licenses from your NEC distributor.

Oracle database

If you choose to use an Oracle database, you must purchase it separately. Oracle 7.3.x versions may be installed on the AIMWorX server computer or on a computer networked to the AIMWorX server. In either case, all AIMWorX clients must have Oracle Client 7.3.x installed. This is done automatically during the AIMWorX client installation.

Licensing

AIMWorX modules are sold and licensed on a per customer basis. You purchase an AIMWorX tier, the AIMWorX modules you need, and the number of seats for each module. Your security key is programmed with these variables and gives you access to the appropriate parts of the installed program.

Seat licenses

Each module you purchase comes with five seats or instances of this module. The license server will let you have five copies of the module active at a time. Starting a sixth copy will cause an access violation error. Purchase groups of five seats for any module from your AIMWorX distributor.

SQL Server database licenses are sold separately, based on the database manufacturer's requirements.

Security

A security program on the server prohibits AIMWorX usage beyond what you have licensed. This program keeps track of the number clients using AIMWorX, the modules you have licensed, and the instances of each module. It interacts with your security key and the application on each client to allow or disallow you access based on the license you purchased and what is currently running.

The security key plugs into the parallel port of the server computer. It is programmed in combination with your serial registration number, so cannot be copied or used on other installations.

Information you need before installing AIMWorX

Before you install any of the AIMWorX modules, you need to know some information about the phone service carrier, the switch, the trunks & routes, the organization, and how you will cost the calls.
AIMWorX supports, but is not limited to:

C/O Line	VPS
DID-incoming only	Conference
Tie line	CAS Line
F/X-Foreign Exchange	Data DM
ISDN	Hot Lines

Configurations

Standalone

This allows the user to only access the AIMWorX applications from the server on which they are installed (Network Printers and email can still be used - Also the freely licensable Alarm Clients can still be used from Network PCs). This includes the AIMWorX Program Disk (1 CD) and the AIMWorX Database Disk (1 CD). The AIMWorX Database Disk provides Microsoft SQL 2000 both Personal and Standard Editions. The Standard Edition is used with Windows 2000 Professional/Server and with Windows NT Workstation/Server. The Personal Edition is used with Windows 98. It is HIGHLY Recommended that you install Standalone AIMWorX Systems on a Microsoft 2000 Professional or Microsoft XP Professional Operating System.

C/S NEC

This is a Client/Server configuration that includes 5 MS SQL 2000 Client Access Licenses. Additional Client/Access Licenses may be selected below. This includes the AIMWorX Program Disk (1 CD) and the AIMWorX Database Disk (1 CD). The AIMWorX Database Disk provides Microsoft SQL 2000 both Personal and Standard Editions. The Standard Edition is used with Windows 2000 Professional/Server and with Windows NT Workstation/Server.

C/S CUST

This is a Client/Server configuration. It does not include a database. The Database must be provided by the customer. Currently supported Databases are Microsoft SQL 2000 (SP1 and above) and Oracle 7.3.4. The customer is responsible for purchasing the database of their choice and the database licenses they desire. This option includes the AIMWorX Program Disk (1 CD).

Call Record and User Capacity

You can purchase AIMWorX in one of six tiers: Basic, Business, Small Corporate, Corporate, Enterprise, and Large Enterprise. These tiers differ based on the number of user records each supports. The smaller tiers can be run as standalone systems, while others operate in a client/server environment only.

Basic tier

The Basic tier manages up to 200 extensions or users. It can be installed as a standalone system or in a client/server environment and works with the *UNIVERGE* NEAX 2000 switch family, the *UNIVERGE* NEAX 2400 switch family, and some non-NEC switches for SMDR collection only. This system supports AIMWorX Manager, call accounting features, and most AIMWorX features and modules.

Business tier

The Business tier manages up to 500 extensions or users. It can be installed as a standalone system or in a client/server environment and works with the *UNIVERGE* NEAX 2000 switch family, the *UNIVERGE* NEAX 2400 switch family, and some non-NEC switches for SMDR collection only. This system supports all AIMWorX features and modules.

Small Corporate tier

The Small Corporate tier manages up to 2,500 extensions or users. It can be installed as a standalone system or in a client/server environment and works with the *UNIVERGE* NEAX 2000 switch family, the *UNIVERGE* NEAX 2400 switch family, and some non-NEC switches for SMDR collection only. This system supports all AIMWorX features and modules.

Corporate tier
The Corporate tier manages up to 5,000 extensions or users. It can be installed as a standalone system or in a client/server environment and works with the *UNIVERGE* NEAX 2000 switch family, the *UNIVERGE* NEAX 2400 switch family, and some non-NEC switches for SMDR collection only. This system supports all AIMWorX features and modules.

Enterprise tier
The Enterprise tier handles up to 65,000 extensions or users. It must be installed in client/server format. It works with the *UNIVERGE* NEAX 2000 switch family, the *UNIVERGE* NEAX 2400 switch family, and some non-NEC switches for SMDR collection only. This system supports all AIMWorX features and modules.

Large Enterprise tier
The Large Enterprise tier handles up to 125,000 extensions or users. It must be installed in client/server format. It works with the *UNIVERGE* NEAX 2000 switch family, the *UNIVERGE* NEAX 2400 switch family, and some non-NEC switches for SMDR collection only. This system supports all AIMWorX features and modules.

The AIM*WorX* Manager

The AIM*WorX* Manager is the foundation of AIM*WorX*. This allows a user to synchronize databases and to control multiple modules via a true single point of entry. It also includes a complete call accounting system offering over 20 reports, each of which can be fully customized and saved for later use. Reports can be scheduled far in advance to print to file (in a variety of formats) or to paper. Reports printed to file can be viewed over the network via a small application loaded on the interested party's computers. Printing times can be scheduled to suit the specific needs of the company.

Integrating AIM*WorX* modules

AIMWorX modules add flexibility to your telemanagement system because you only purchase the modules that you need. Nearly all of them interface with AIMWorX Manager. Some are accessed directly through a tab on the Users screen, while others have separate interfaces, but use the call accounting database. AIMWorX modules seamlessly integrate with each other. To ensure this integration, certain modules should be purchased together. Some modules should not be paired.

AIMWorX works optimally if these modules are paired:
- Cable Manager and Asset Manager
- Work Order and Trouble Ticket
- Asset Manager and Billing Manager
- NEAX SPE and Voice Mail SPE
- NEAX SPE and Auth Code Manager
- NEAX SPE, 9-1-1 OSN, and Alarm Manager
- AIMWorX Manager and Multi-site
- Hospitality Links and Voice Mail SPE
- Traffic Analysis and NEAX SPE

These modules are mutually exclusive and should not be purchased together:
- Interactive Directory 2000 and Interactive Directory 2400
- Interactive Directory 2000/2400 and Message Center

Work Order should have these three modules at a minimum to work:
- NEAX SPE
- Asset Manager
- Billing Manager

Work Order will best enhance your system's capabilities when combined with these modules:
- Trouble Ticket
- 9-1-1 SPE
- 9-1-1 OSN
- Interactive Directory
- Auth Code Manger
- Cable Manger
- Voice Mail SPE

9-1-1 SPE

9-1-1 SPE is a module that lets you download specific user location information directly to the phone company's ALI system. It appears as a tab on the Users screen in AIMWorX Manager and exports the 9-1-1 user database in NENA format. The 9-1-1 SPE is not involved at the time of an emergency; it supplies user location information to the ALI system, which then supplies it to the authorities when needed. 9-1-1 On Site Notification is a feature of Alarm Manager that will notify your own security department with the same information when someone places a 9-1-1 call.

Alarm Manager

The Alarm Manager feature of AIMWorX provides a service that collects and forwards 9-1-1 OSN, Traffic, Collector, Auth Code and other alarm notices to Alarm Clients. Alarm Clients must be networked to AIMWorX, but do not need to be AIMWorX clients. Through the Alarm Manager you can define which clients receive which alarm notices, and what happens if the alarm is not acknowledged.

Asset Manager

The Asset Manager module of AIMWorX is a way to track stock inventory once it becomes an asset. Asset Manager lets you record all assets and assign them to categories and subcategories. It track warranties, vendors, and manufacturers for assets and lets you bill them to a user, department, or customer. This program works in conjunction with the other AIMWorX modules and has extensive import/export capabilities in order to share data with your current accounting and inventory programs.

Auth Code Manager

Auth Code Manager is in charge of tracking and monitoring authorization codes as well as granting permissions at the time of a call. Use authorization codes to define limits and access for specific users. Auth Code Manager interfaces to the AIMWorX database for user information and appears as a tab on the user screen.

Billing Manager

Billing Manager is a module of AIMWorX that provides a way to charge a customer, department, or company for calls, equipment, and services. It is not a full-blown accounting system, but a subsystem of AIMWorX that will export phone related charges in formats acceptable to your organization's current accounting package. This module works with the AIMWorX Manager, Asset Manager, and Work Order to provide a billing solution for your telemanagement needs.

Bill Reconciler

Bill Reconciler compares monthly billing statements from the telephone service carrier to CDR generated by AIMWorX. You can also save the results from this comparison and create reports using this cost, your cost, or the billed cost, or the differences between them.

Cable Manager

Cable Manager lets you maintain information on the cable and wire layout of your facility. It lets you perform powerful end-to-end searches and recommends a cable path, lets you review it, then reserves it. You can define each down to the levels of rooms and closets in a drag and drop graphic representation. On other screens, assign switch nodes and ports, frames, blocks, pairs, cross-connects, locations, and include descriptions. Cable Manager complies with EIA/TIA-568 A&B wiring standards and the EIA/TIA-606 Wire Records Keeping procedures recommended by the International Standards Body.

Custom SPE Writer

Custom SPE Writer lets you create AIMWorX SPE modules for applications outside of AIMWorX. Use your own development tools to create an interface using commands that talk to the AIMWorX Arena. Each application will then appear as a new tab on the Users screen and you can populate the external database from inside AIMWorX.

Hospitality Links

Hospitality Links combines the power of your Property Management System (PMS), the flexibility of your *UNIVERGE* NEAX 2400 IMX/IPX Hospitality switch, and the control of AIMWorX into one solution for your Hospitality environment. This module lets you manage your administrative call accounting while formatting and forwarding guest calls to your PMS for billing. The AIMWorX Suite Master application is a part of Hospitality Links and lets you manage the Flexible Suiting feature of the *UNIVERGE* NEAX 2400 IMX/IPX Hospitality switch.

Interactive Directory

Interactive Directory is an electronic phone book and more. It provides a directory of user information including extension, fax number, phone number, emergency number, e-mail address, and twelve custom fields of information. The directory can be published from AIMWorX Manager or you can enter user information directly through the Interactive Directory Manager program.

NEAX 2000 SPE

NEAX SPE is a module that lets you manage some of the switch Maintenance and Administration Terminal (MAT) functions from your AIMWorX computer. This program is a typical example of the concept of ‘single point of entry’. Once the user information is in the AIMWorX system, you can program the switch with information to assign phones, extensions, and other administrative duties. This is all done from a single screen in AIMWorX Manager.

Traffic Analysis

Traffic Analysis intercepts the *UNIVERGE* NEAX 2000 IPS statistics, such as peg counts, from the PBX's MAT port and uses them to analyze your telecommunications needs. This module shows your single or multi-switch system in a graphic layout. It provides a graph of call traffic between PBXs and lets you zoom in on specific sections to assess usage of specific paths in 30-minute segments.

Trouble Ticket

Trouble Ticket simplifies the trouble reporting, resolution, and tracking process. It lets you see a database of telecom problems and their associated solutions with a simple character search. Suggest simple solutions over the phone or create work orders, on the same screen, for the more involved problems.

Voice Mail SPE

Voice Mail SPE is an AIMWorX module that lets you program the voice mail system from a tab on the AIMWorX Manager user's screen. This interface lets you avoid the time needed and mistakes made by entering the duplicate information into two systems. The AIMWorX Voice Mail SPE currently supports the Octel Overture 250, Baypoint/Centigram C70, 120I, 120R, 120S, or 640, Captaris/AVT CallXpress, and NEAXmail AD-40 and NEAXmail AD-120 voice mail systems.

Work Order

Work Order lets you create work orders that automate commands from all the other AIMWorX modules. It uses a separate user interface that lets you schedule work orders, manage technician's schedules, track progress through completion, and manage inventory and billing.

Additional Reference Material

All additional documentation is included on the CD ROM. Also included on the CD-ROM is a copy of Acrobat Reader to view the documentation.

Chapter 19 Voice Messaging Systems

Voice Mail Integration

Voice mail allows callers to leave a recorded message, in their own voice and exact words, for another individual or group of individuals. With a voice mail application, people can communicate precisely and personally without having to be on the same telephone line at the same time. Voice mail applications allow communication to take place without endless games of telephone tag. Research has shown that only one-third of all business calls achieve direct contact, yet 40% of all messages are one way messages that do not require dialogue. Voice mail saves time by allowing you to leave a message immediately. Voice mail bypasses the problems of time zones, after-switchboard hours, and other obstacles to communication in this fast-paced business world. Voice mail allows the people in your office to better manage their communications and time.

Voice mail features can be used by accessing the VMS directly from an extension. If a station sets its call forwarding destination to the VMS, calls to this station are connected to the VMS, and the messages can be registered according to the VMS instruction. In addition, the Message Waiting lamp of the station can be turned on automatically by the VMS.

The following voice mail systems to integrate to the *UNIVERGE* NEAX 2000 IPS system.

NEAXMail AD-120: The AD-120 is available in 4 to 72-port configurations and offers both analog and digital station port integration. . The AD-120 may cluster up to 5 72-port systems supporting a total of 360 ports. The AD-120 also provides a range of capabilities, from voice messaging to Unified Messaging.

***NEAXMail AD-64*:** The AD-64 is available in 4 to 96-port configurations and is digitally integrated to the NEAX 2000 IPS PBX. The NEAXMail AD-64 comes as a Level I or Level II system and offers a “25 free seat license of TeLANophy” along with other optional features.

NEAXMail IM-16 LX: A card type voice processing system with various features and is mounted into the LT slot of the PBX. The basic system of the NEAXMail IM-16 LX provides 4 or 8 ports of voice mail. Adding an additional expansion card provides 4 or 8 additional ports yielding a maximum of 16 voice mail ports.

Service Conditions

1. The system transfers only DTMF signals to the connected VMS. It cannot transfer dial pulses to the system.

2. Messages can be retrieved from any Multiline Terminal, DTMF telephone, or the Attendant Console, but not from DP telephones.

3. When the calling party is connected to the Voice Mail System, only DTMF signals can be sent to the VMS for registering a message. DP telephones cannot be used.

4. Stations can set Call Forwarding - All Calls, Call Forwarding - No Answer, and Call Forwarding – Busy Line to the VMS. The system sends out a mail box number to the VMS. Calling a station that has Call Forwarding set to the VMS is automatically answered by the VMS.

5. The DTMF signal pause, Inter-Digit Pause, and DTMF signal width of the station number automatically sent out to the VMS from the system are as follows:

    a. Pause: Variable from 1 second to 12 seconds in 1 second increments
    b. Inter-Digit Pause: Variable from 32 milliseconds to 240 milliseconds
    c. DTMF signal width: Fixed at 64 milliseconds or 128 milliseconds

6. A special number of up to 4 digits (including an Inter-Digit Pause) can automatically be added, both before and after, to the station number that is sent to the VMS from the system. This can be used for a variety of identification codes as required. Two types of Inter-Digit Pauses can be set per system. One is fixed at 1.5 seconds; the other is programmable from 1.5 seconds to 16 seconds.

7. The VMS can control the Message Waiting Lamp of the Station set by using the Message Waiting/Message Reminder feature. The retrieval access code for Message Waiting/Message Reminder is variable and can be set from 1 to 4 digits, in system programming.

8. When all VMS ports are busy (assuming the following condition):

    Station-A (or outside party)....Calling Party
    Station-B.Called Party
    Station-B sets Call Forwarding - All Calls/Busy Line/No Answer to the VMS.
    Station-A (or outside party) makes a call to Station-B.

    a. Call Forwarding - All Calls :
        a. Station A hears reorder tone.
        b. Outside party hears busy tone.

    b. Call Forwarding - Busy Line
        a. Station-A hears busy tone, and can set any busy service to station-B.
        b. Outside party hears busy tone.

    c. Call Forwarding - No Answer
        a. Station-B continues to ring until the VMS becomes idle regardless of whether the predetermined time for Call Forwarding - No Answer has elapsed. When the VMS becomes idle, Station-A is connected to the VMS.

    d. Direct access to VMS
        a. If station A or outside party accesses the VMS directly, the outside party hears busy tone and station A can set call back to the VMS.

9. Multiple Call Forwarding: When the final destination for any combination of Multiple Call Forwarding is the VMS, calls can be transferred to the VMS. The first forwarded station's number is sent to the VMS. For example, a call is received by Station A, which is forwarded to Station-B, which is forwarded to Station-C, which is forwarded to VMS. The number of Station-A is sent to the VMS.

10. Ringing Transfer to an Attendant via the VMS: The system allows the VMS to transfer the station or outside party to the Attendant and releases before the Attendant answers.

11. When the VMS is recalled, by transferring the call to an unanswered station, the system may be programmed to send the recalling extension number to the VMS.

12. A maximum of 26 digit extension numbers including DTMF signal after the VMS answered, can be programmed to a One Touch key of a Multiline Terminal.

Message Center Interface (MCI)

This feature provides an interface to the Voice Mail System (VMS), which can send Message Waiting lamp and other control data to the system. The Message Center Interface (MCI) can provide the following operations:

1. When terminating the call to the VMS, the system sends call connection status information to the VMS through the MCI.

2. The VMS sends the Message Waiting Lamp on data to the MCI.

3. The system, upon receiving this control data from the MCI, illuminates the Message Waiting lamp of the corresponding station.

4. The VMS, upon receiving retrieved message information, will send the Message Waiting lamp control data requesting the system to extinguish the Message Waiting lamp of the corresponding station.

5. The MCI interface is a half-duplex, EIA-RS232C, asynchronous serial data link that operates under a specific message protocol and format. The characteristics of the link, the protocol used, and the message text are shown below.

ITEM	DESCRIPTION
Data Rate	1200, 2400, 4800, 9600 bps asynchronous, software selectable
Operating Mode:	Half duplex
Electrical Interface Characteristic:	EIA RS-232C electrical standard interface
Signal Form:	EIA RS-404
Interface Distance:	Max. 15 m (49.2 ft.) between PBX and VMS (without modem) **Note:** *When modems are used, full duplex asynchronous type modems are required.*
Word Framing:	10 bits (1 start, 7 data, 2 stop)
Parity VRC:	No parity, even parity, odd parity; selected by PBX system data
Frame Contents:	US ASCII 7-bit codes
Control:	Contention
Priority Sequence:	Primary station: PBX Secondary station: VMS

MCI Service Conditions

1. The UCD or Station Hunting feature is usually provided with the VMS station.
2. One RS-232C port on the MP card or the Application Processor (AP00) is required to make a data link with a customer supplied VMS.
3. Messages can be retrieved from any Multiline Terminal, DTMF telephone, or the Attendant Console, but not from DP telephones.
4. The MCI is available to a direct call or a forwarded call from a station/trunk/Attendant to the VMS. For details of the connecting patterns, refer to the Installation Procedure Manual/Feature Programming Manual.
5. Stations can set Call Forwarding or Split Call Forwarding - All Calls, No Answer, and Busy Line to the VMS. The system sends out incoming call information to the VMS. A call to a station that has Call Forwarding set to the VMS is automatically answered by the VMS.
6. The MCI can control the LCD display of a Multiline Terminal for “MESSAGE” Indication. The number of messages is not displayed.
7. When the Message Waiting lamp control is activated with the MCI, the lamp control from the following equipment will not be provided:
    - From the Property Management System (PMS)
    - From the Station (by dialing the access code)
    - From the Direct Station Selection (DSS) Console
    - From the Hotel/Motel Attendant Console
    - From the Hotel/Motel Front Desk Instrument
    - From the Attendant Console
8. Only one system should be programmed (via system programming) to control Message Waiting lamps through the CCIS network.
9. The system controls Message Waiting lamps normally when the time interval between messages is a minimum of 350 msec. or more.
10. When the VMS interface line does not answer, all of the messages are sent out from the I/O port of the MP/AP00.
11. If the VMS is not ready for information receiving (Busy Status), the AP00 can temporarily store up to 16 call records in its internal memory. If the maximum of 16 call records is stored and a 17th is generated, the system will write over the oldest stored record.
12. When the RS port on the MP card is used for the data link to the VMS, the MP can store up to 15 call records. If a 16th call record is generated when the MP stores 15 call records, the system will write over the oldest stored record.
13. The Voice Mail Integration (In band) feature can be combined with voice mail through the MCI in the system. The Voice Mail Integration (In band) feature and MCI feature can coexist in one system and either can be selected per VMS (VMS station number) by system programming.
14. When terminating a call with the ANI information to the VMS through the MCI, the system can send the ANI information to the VMS, if required. This is not available through CCIS interface.

NEAXMail AD-120

The NEAXMail AD-120 Product offers the following two choices for system configuration:

- **Voice Mail Only** - provides voice mail only services. Voice mail only systems are shipped with all software preloaded.

- **Unified Messaging** - provides full voice messaging capabilities plus an unlimited seat license for the View Mail for Outlook (VMO) Unified Messaging client software. Third Party Fax Integration and Text-to-Speech are available as optional components. Unified Messaging systems are shipped without software pre-loaded. All software must be loaded on-site in order to effectively join the customer's existing Windows network.

The AD-120 offers both analog and digital station port integration.

- **PBX Integration** – All AD-120 systems include the NEAX Digital and Serial MCI switch integrations. The following components provide integration support for Cisco Call Manager (Version. 2.4 or above) and SMDI Serial systems. T1 integration is not currently supported with Dual IP/PBX Support. Enhanced NEAX OAI Integration must be ordered for each AD-120 in the cluster or whenever Live Record is desired in an analog port configuration.

System Architecture

- **Microsoft® Exchange® 2003 Server** ships with AD-120 S/W Port Licenses that include Exchange. Associates may elect to purchase Exchange through NEC or to provide their own copies.

- **Microsoft SQL® 2000 Server** is the underlying database for AD-120. SQL is provided either in MSDE or Server Standard 2000 Runtime Edition license versions. MSDE is provided with S/W Port Licenses from 4 to 28 ports. Port Licenses above 28 ports require the purchase of SQL Server Standard 2000, Processor License. As a "Processor License," SQL must be purchased in quantities equivalent to the number of CPUs in the server. Therefore, a Dell PE2800 with dual processor's, requires two licenses. NEC Authorized Associates may elect to provide their own copies if desired.

Voice Cards

Voice boards must be purchased for all NEAXMail AD-120 orders.

Note: *Systems 4 to 20 ports require a minimum of 2GB RAM, 24 ports to 32 require a minimum of 3GB RAM, above 32 ports require 4GB RAM. Systems above 48 ports require dual processors. Systems with 32 ports or higher must also purchase the proper quantity of SQL Standard Server 2000 Runtime Edition.*

Description
4 Port - D/42-NE2 Digital Voice Board
4 Port - D/4 PCI Analog Voice Board
4 Port – D/41 JCT-LS Analog Voice Board
8 Port – D/82 JCT-U Digital Voice Board
12 Port - D/120JCT-LSU Analog Voice Board
24 Port – D/240PCI-T1R2 T1 Voice Board

Level II Server Software License with Exchange 2003

A NEAXMail AD-120 Software License includes:

- NEAXMail AD-120 Version 2.8 software
- SQL MSDE version (4 – 28 ports only)
- Exchange 2003 Server (Voicemail Runtime edition)
- Designated Voice Port license
- Unified messaging seats or Voice mail seats (240 per 4 ports)
- System key & documentation

Microsoft Exchange 2003 Server (Voice Mail Run-time edition) satisfies the requirement for Exchange 2003 Client Access Licenses (CALs) for all 'voice mail only' users on the system. CALs are still required for all "unified messaging" subscribers and may be purchased from NEC or provided through a third party.

NEC Authorized Associates may elect to provide their own server platform and third party software (i.e., Microsoft Windows 2003, PCAnywhere, Exchange 2003, and SQL 2000). The NEC Associate assumes all responsibility for providing a Microsoft certified Windows 2003 server. NEC's warranty excludes any Associate supplied computer platform or software used in conjunction with the NEAXMail AD-120 system.

Unified Messaging license and Voice Mail only license provide 240 seat licenses per 4 ports.

Voice boards must be purchased through NEC. AD120 Port License are ordered as specified in the chart below.

Voice Mail only Software Licenses
AD120 VM Parent, Exchange 2003, 240 Subscribers, per 4 Ports
AD120 VM Parent, No Exchange, 240 Subscribers, per 4 Ports

Unified Messaging Software Licenses
AD120 UM Parent, Exchange 2003, 240 Subscribers, per 4 Ports
AD120 UM Parent, No Exchange, 240 Subscribers, per 4 Ports

Third Party Software

Description
AD-120 Third-Party S/W (Win2003 + PCAnywhere)
SQL Server Standard 2000 Runtime Edition (per CPU)

If you wish to purchase the Windows 2003 operating system software package for a new system order from NEC, you MUST also purchase a Level I or II platform.

Windows 2003 is installed on AD-120 voice mail only platforms. If you elect to purchase only the AD-120 software license and supply your own platform, you must provide Windows software.

Microsoft SQL Server Standard 2000 Runtime Edition is required for AD-120 configurations of 32-ports or greater. One copy of SQL Server Standard 2000 Runtime Edition is required for each processor in the AD-120 platform. For example, a 32-port AD-120 on a Dell Power Edge 2800 (with Dual CPUs) would require a quantity of two licenses.

Client Access Licenses (CAL)

Available CALs
AD120, Exchange 5.5 CAL (per seat)
AD120, Exchange 2000 CAL (per seat)
AD120, Exchange 2003 CAL (per seat)

Exchange Server 2003 (Voicemail Runtime Edition) included with the AD-120 server licenses (as designated above) will exempt all voice mail only users from CAL requirements. CALs are still required for all unified messaging subscribers and may be purchased from NEC or provided from a third party.

If AD-120 Software License is purchased without Exchange Server 2003 (Voicemail Runtime Edition), Exchange Server 2003 (Standard Edition) is required to run AD-120. Exchange Server 2003 (Standard Edition) requires CALs for all users on the AD-120 system, whether they are voice mail only or unified messaging subscribers. CALs maybe purchased from NEC or provided from a third party.

Platform Components

The NEAXMail AD-120 provides separate platform components (i.e., individual AD-120 Platforms, Software Licenses, Voice Boards and Third Party Software), allowing NEC Associates a greater flexibility in meeting their customer's requirements. NEC Associates may choose to order "software & voice boards only" or a combination of software, voice boards and a Level I or II platform. A complete AD-120 ***system*** would include a Level I or II Platform, AD-120 Server Software License, voice boards and third party software ordered together.

- If a Unified Messaging Software License is ordered with a Level I or II platform, the system will ship without pre-loading of software. This is because installation of the AD-120 into an existing network requires the operating system software (and Exchange) to be loaded while connected to that Local Area Network (LAN).
- RAID hard disk drive redundancy (Level II) may only be added at the time of purchase and is not field-upgradeable. The optional RAID package provides four (4) additional hard drives for a total of six (6) hard drives in the Level II server and doubles the internal storage capacity.
- Rack Mount Kits allow the Level II Dell Servers to be mounted in a standard 19" rack.

Description
AD120, Level I, Dell GX280
Dell Level II PE2800 Level II Tower
Dell Level II, 6 Drive RAID Array, (PE2800 only)
Option Dell Level II Rack Mount Kit (PE 2800 only)
Secondary CPU (Dell Level II PE2800, Xeon 2.8GHz)
CPU Option (Dell Level II PE2800, Xeon 3.6GHz)
1GB Memory addition, (Dell GX280 only)
1GB Memory addition, (Dell Level II PE2800 only)
2GB Memory addition, (Dell Level II PE2800 only)
17 Inch Color Monitor
External USB Modem

Optional Software Feature Packages

ActiveNet, AMIS, VPIM and Third Party Fax Server Integration options may only be installed by an NEC Certified NEAXMail AD-120 engineer or an AD-120 technician with the assistance of a qualified MCSE.

Description
ActiveNet for Unity/AD-120 (per 1 node)
AD120, AMIS Analog Networking (per server)
AD120, VPIM Digital Networking (per server)

E-Mail Reader (Text-to-Speech)

The AD-120 offers a Text-to-Speech (TTS) Reader. The TTS read text in English only. TTS is required for Outlook Calendar Access and is not available on Voice Mail Only systems.

Capacities: Minimum of 2, Maximum of 20 (RealSpeak) simultaneous sessions per system.

RealSpeak Note: *1 to 3 sessions require a total of 2GB RAM, 4 sessions require a total of 3GB RAM, Above 5 sessions require 4GB RAM. Above 3 sessions requires a Level ll platform. Above 8 sessions require Dual Xeon 2.8 GHz processor option*.

Description
RealSpeak Text To Speech Reader, 1 Session

Third Party Fax Server Integration

The following fax servers are supported: Omtool Fax Sr. v.3, Interstar Technologies LightningFax v.5.5 or higher, Fenestrae FAXination v.4 or higher, RightFax v.6 or higher, Optus FACSys v.4.5 or higher, FAXCOM for Exchange version 6.19 or higher and Esker FaxGate v.7 or higher.

PBX Integration Software

All AD-120 systems include the NEAX Digital and Serial MCI switch integrations. The following components provide integration support for Cisco Call Manager (ver. 2.4 or above) and SMDI Serial systems. T1 integration is not currently supported with Dual IP/PBX Support. Enhanced NEAX OAI Integration must be ordered for each AD-120 in the cluster or whenever Live Record is desired in an analog port configuration. Load Balancing is required when you want to cluster 2 AD120's and have them automatically load balance the calls.

Description
AD120 Dual Integration (IP + PBX)
AD120 SMDI Serial Integration
AD120 Enhanced NEAX OAI Integration
AD120 Load Balancing

Platform Specifications

NEAXMail AD-120 Specifications		
Platform Level	**Dell Level I**	**Dell Level II (PE2800)**
CPU Type/Speed	Pentium 4 2.8Ghz	Xeon 2.8Ghz
Dual CPU's	N/A	Optional (2.8 or 3.6Hz)
MB RAM Installed	2GB	2GB
PCI Slots	3 (all 5-Volt)	5 (one 5-Volt & Four 3.3-Volt)
Chassis Style	Mini-Tower	Tower
Rack-Mount	Not Available	Optional
Maximum Voice Port Capacity*	16 Analog, 16 Digital	72 Analog or 64Digital
Hours of Storage	Approx. 530	Approx. 1200 or 2400 with RAID
Hard Disk Drive Size	Dual 40GB SATA	Dual 36GB SCSI
RAID Array Drive Size(s)	N/A	Optional 6 x 36GB
Tape Backup	Optional SCSI	Optional
Expansion Chassis	N/A	Optional (PCI)
Redundant Power Supplies	N/A	Standard

NEAXMail AD-64

The NEAXMail AD-64 voice/unified messaging system offers a powerful centralized message solution to meet the communications needs of your business. You can manage your communications over the telephone or from your personal computer (PC) using the Microsoft® Outlook®, Lotus® Notes®, and GroupWise® Inbox. The NEAXMail AD-64 offers advanced unified messaging, voice mail and automated attendant functionality, helping your organization become more responsive than ever. Through its suite of unified messaging and desktop call management applications, NEAXMail AD-64 provides you with one of the most complete unified communications solution available.

These unified communication applications integrate the NEAXMail AD-64 with your Local Area Network (LAN) to give you control over live telephone traffic and messages – voice, fax* and e-mail – all from your desktop PC. The ViewMail® for Microsoft Messaging module, for example, works with Microsoft Exchange® to allow you to access your messages from one familiar inbox. ViewMail® for Lotus Notes and ViewMail for GroupWise are also offered. In addition, the ViewCall® Plus module enables you to handle your live telephone traffic from your desktop PC and also works with the applications you use every day such as your Outlook contact list and your Access® database.

The NEAXMail AD-64 provides a foundation for future growth and customization. Built upon the Windows 2003 Server platform, the NEAXMail AD-64 leverages the power, reliability and scalability of the Microsoft Windows® 2003 operating system. By using the new Windows-based system administration console, the NEAXMail AD-64 can be administered from any Windows PC on your LAN. This new console is intuitively designed and is accessible via an IP connection, making system administration easier and more efficient. The Windows 2003 Server also provides many of the disk maintenance utilities you already know, eliminating the need to learn new software interfaces.

The NEAXMail AD-64 is available in 4 to 96-port configurations with analog or digital integration to your NEAX® PBX. The NEAXMail AD-64 may be purchased on either a Level I or Level II platform. The specifications for these platforms are listed below; however system specifications are subject to change without prior written notice.

Level I/Level II Platform Specifications

Level I/Level II Specifications		
Platform Level	**Level I (Dell GX280)**	**Level II (Dell PE2800)**
CPU Type/Speed	Pentium 4 2.8GHz	Xeon 2.8GHz
Dual CPUs	N/A	N/A
RAM Installed (max)	512GB (4GB)	512GB (6GB)
PCI Slots (voltage)	3 slots total – all are PCI (5-Volt)	5 slots total - 4 PCI-X (3.3Volt) and 1 PCI (5Volt)
Chassis Style	Mini-Tower	Tower
Rack-Mount	Not Available	Optional
Cabinet Color	Black	Black
Max. Voice Port Capacity **Note**	48 Analog or Digital	96 Analog or 64Digital
Storage Hours	Approx. 1650	Approx. 3200 or 6400 w/RAID
Hard Disk Drive Size	40GB SATA	36GB SCSI
RAID Array Drive	N/A	3 x 36GB
Tape Backup	Optional SCSI	Optional
Redundant Power Supplies	N/A	Standard
Max. Wattage	250 Watts	730 Watts
Heat dissipation	853 BTU/hr	3100 BTU/hr
Chassis Height	16.7 inches (42.5 cm)	17.5 inches (44.5 cm)
Chassis Width	7.1 inches (18.1 cm)	9.125 inches (23.0 cm)
Chassis Depth	17.6 inches (44.7 cm)	24.54 inches (62.3 cm)
Weight	28 lbs (12.7 kg)	90 lb (40.9 kg), max. configuration
Operating Temperature	50° to 95°F 10° to 35°C	50° to 95°F 10° to 35°C
Storage Temperature	-40° to 65°C -40° to 149°F	-40° to 65°C -40° to 149°F
Relative humidity	20% to 80% (non-condensing)	20% to 80% (non-condensing)

Note: *may require Expansion Chassis*

Additional Hardware Components

Expansion Chassis

Available in a 7-slot PCI form factor, the Expansion Chassis provide additional card slots for voice or fax cards.

Note: *The Expansion Chassis can be added to any NEC-provided platform. NEC Associate assumes all responsibility of PCI Expansion Chassis compatibility with all other "Associate-provided" hardware. Up to two expansion chassis may be connected to a Level II (or equivalent) server platform.*

ActiveFax Board

The VFX/41JCT-LS fax card provides 4 fax ports on a single PCI Card. ActiveFax software licenses are not included and are sold separately.

Tape Backup

The tape backup option includes a Tape drive, 1 DAT tape, software, cleaning cartridge and documentation.
Note: *Level I systems require the addition of a SCSI adapter card, which requires 1 open PCI card slot.*

UPS (Uninterruptible Power Supply)

In the event of a power failure, the power supply will instantly switch to its internal battery bank. This keeps the NEAXMail AD-64 in continuous operation and allows the user to work and save valuable data before the platform shuts down.
Note: *NEC strongly recommends the use of a UPS for all NEAXMail AD-64 systems.*

Optional Software

Operating System and Utilities Software

The Third Party software includes:

- Windows 2003 Server software (Runtime Edition)
- PCAnywhere 11.5 or higher.

Notes:

- *Windows 2003 Server software (Voicemail Runtime Edition) and PCAnywhere may be purchased from NEC or through a third party provider. If you choose to purchase the Third Party Software from NEC you MUST also purchase a PC/Server platform or in conjunction with a NEAXMail AD-64 Upgrade order.*
- *If you choose to provide your own Windows 2003 server and PCAnywhere software, it is the Associates responsibility to install Windows 2003 server software, PCAnywhere and voice messaging software.*
- *Windows 2003 Server (Voicemail Runtime Edition) may only be used to run the NEAXMail AD-64-related applications on the NEAXMail AD-64 platform.*

Optional ActiveFax Software

Maximum ActiveFax Software capacity is 8 in a single system. Fax card(s) (NEC#0161975) are not included and are sold separately.

Unified Messaging

A NEAXMail AD-64 Port Software License includes 25 seats of ViewMail/ViewMail for Microsoft Messaging/ViewMail for Lotus Notes/ViewMail for GroupWise and 25 seats of ViewCall Plus.

Additional seats of these applications may be purchased in 5, 50 or unlimited bundles. The Unlimited package provides up to 1,250 simultaneous sessions of ViewMail/VMM/VML and ViewCall Plus.

ViewCall Plus requires either TAPI or TSAPI services for enhanced mode options. Please consult your NEC Sales Engineer with CTI related questions.

Feature Descriptions

Analog Voice Card support offers greater flexibility in the overall NEAXMail AD-64 configuration. Analog cards allow for a maximum port capacity of 96 ports and allow previous NEC Repartee® OS/2 customers to upgrade to the NEAXMail AD-64's Windows® 2003 operating system.

Announcement of number and length of messages NEAXMail AD-64 announces how long messages last; for example, "You have three new messages totaling three minutes, twenty seconds. Would you like to hear them?"

Archiving The message archive feature allows subscribers to save messages longer than the preset system parameter.

Automated Speech Recognition (ASR) provides subscribers with the ability to navigate their mailbox options using speech commands. By speaking simple commands into any telephone, you can navigate your voice mailbox and change personal options.

Caller ID information is accessible via the telephone. When a user checks a message that has Caller ID information associated with it, the number will appear on their Dterm's LCD display. Caller ID can also be retrieved through the soft key menu "CID" or through ASR by saying "Caller ID." This feature requires the Digital Dterm voice card(s) (i.e., D/42-NE2, D/42JCT-U or D/82JCT-U).

Call Return is a feature that allows a subscriber to callback the sender of a voice message based upon the Caller ID captured during that message. This feature requires the Digital Dterm voice card(s) (i.e., D/42-NE2, D/42JCT-U or D/82JCT-U).

Date and time stamp. At the end of every message, NEAXMail AD-64 indicates how long ago a message was sent, providing subscribers with accurate information about their messaging activity.

Greetings Each subscriber mailbox can have up to three greetings: standard, busy, or alternate. Greetings can be rerecorded or switched by the subscriber via their Dterm set, touchtone phone or through the Mailbox Manager application. If you do not record greetings in your own voice, the system plays greetings that include your recorded name or your extension.

- **Standard greeting** - plays when your extension is unanswered. A typical standard greeting is: "Hello, this is Pat Green. I am not at my desk right now. Please leave a message."
- **Busy greeting** - plays when your extension is busy. A typical busy greeting is: "Hello, this is Pat Green. I am on the phone now. Please leave a message." For callers to hear your busy greeting it must activated.
- **Alternate greeting** - plays for special occasions, such as a vacation. A typical alternate greeting is: "Hello, this is Pat Green. I am out of the office today and will return tomorrow." When your alternate greeting is active, the system plays it instead of any of your other greetings.

Guests A guest is a person who is hosted by a particular subscriber and given a "guest mailbox." Guest privileges allow a person to communicate with their host subscriber through voice messages.

Hacker Prevention Features are as follows:

- Mandate a default password on each new box so they are not left "unprotected".
- Allow system administrators to change the default security code via the administration console.
- Force new subscribers to change the default security code upon first-time enrollment.

Hospitality is an optional special guest messaging service, which offers room guests a shortened system conversation for ease of use. This package includes 5 guest languages and PMS Integration.

Last in-First out / First In-First out (LIFO/FIFO). System administrator can set LIFO/FIFO for entire categories of messages (e.g., all messages from other subscribers would be played back in either LIFO or FIFO order; message playback no longer grouped by subscriber name).

- **Message order** allows users to hear messages in the order most convenient.
- **Playback order** allows the system administrator to set playback order separately for new messages and old messages for maximum flexibility and convenience.

Live Monitor is a new digital feature that lets a subscriber listen to a caller leaving a message in their voice mailbox. The subscriber can also pick up the call during the recording to connect. Live monitor is initiated by pressing a line key on a Dterm® Series (III or higher) phone. This feature requires the Digital Dterm voice card(s) (i.e., D/42-NE2, D/42JCT-U or D/82JCT-U).

Message editing by both inside and outside callers With message editing, callers can edit all or part of a message they've just recorded—listen to the message, add to it, delete it, re-record it, etc.

Message-return receipts and receipt summaries. Return receipts give subscribers detailed information about each message he or she sends, including exactly when the recipient opened the message. For a group message, return receipts tell when everyone in the group opened the message (for dispatch groups, it identifies the person who heard the entire message and when). Receipt summaries quickly tell the user whether or not a particular subscriber opened any or all of the messages sent and whether he or she left the sender any messages.

Message waiting notification The NEAX PBX's message waiting indicator is activated when a subscriber has new messages. This may be the extension's Message Waiting Lamp or Stutter dial tone.

Multiple subscriber interfaces. Individual subscribers on the same system have a choice of using the easy 1 for Yes, 2 for No interface or the Menu options interface.

- **1 for Yes, 2 for No interface.** Using the "1 for Yes and 2 for No" option, users access all functions by responding to simple yes-or-no questions.
- **Menu options.** NEAXMail AD-64 offers a traditional voice mail menu interface, as well as the yes-and-no interface. Menu options make it easy for experienced users to access all functions quickly.
- **Quick keys.** Yes-and-no interface users can utilize the 4 through 7 keys to jump to specific options or features.

Outside caller group messaging Outside callers can leave a message for a group of subscribers instead of a single subscriber. Messages can be left to either a public or private group and will follow the delivery method assigned to the group.

Personal Mailbox Each subscriber within an organization can have a private voice mailbox. The NEAXMail AD-64's internal database provides up to 65,000 voice mailboxes.

Private Distribution Lists. Subscribers may have an unlimited number of private distribution lists with an unlimited number of members per list. Only subscribers may send a message to their private lists. Private Distribution Lists are configured with one of two delivery methods:

- **Broadcast Delivery.** All members of the list hear the message when they listen to their messages
- **Dispatch Delivery.** Only the first list member to listen to a message receives it. The message is then automatically deleted from all remaining list members' mailboxes.

Receipt sent for messages saved as new. Subscribers requesting message receipts will receive a receipt as soon as the recipient of a message listens to any part of the message, even if the recipient saves the message as new. Once the recipient has opened a message, the sender cannot cancel the message or change its special delivery options.

Redirect to Multiple Subscribers with one Introduction (RSI). This feature gives users the ability to redirect messages to numerous users without having to record an individual introduction for each recipient.

Rewind, pause, and fast forward. While listening to a message, users press "7" to move backward, "8" to pause, or "9" to skip forward in the message. When recording a message, users can press "8" to pause.

Special delivery options Users can mark messages with one or more of the following special delivery options: urgent, private, future, and return receipt requested. Subscribers can also change or add to the special delivery settings of a message after it's sent.

- **Future.** The sender can mark the voice message for delivery at a later date and time up to one year later.
- **Private.** The recipient of a "private" message cannot redirect the message to anyone else.
- **Return receipt requested.** The sender of the message is told when the recipient hears the message. Subscribers can request return receipts for messages sent to individual subscribers or to groups.
- **Urgent.** Messages marked with "urgent" will be heard first, before regular messages. Urgent messages from other subscribers and guests are played before urgent messages from outside callers. The system also tells subscribers how many new messages are marked urgent.

Speed control during message playback. Subscribers can slow down or increase the speed of each message during playback using the telephone keypad.

Subscriber-controlled groups Subscribers can create, name, add, and delete subscriber groups from their telephone.

Subscriber-controlled message delivery Messages can be delivered to any telephone—home, work extension, pager, or mobile telephone. Each subscriber can have up to four separate telephone numbers and dial-out schedules. The message delivery options can be sequential for each telephone number and programmed by the subscriber. A subscriber can use any touchtone telephone to turn message delivery on or off, or change the delivery times.

Subscriber language The system manager can set which language each subscriber hears. Subscribers will hear prompts in that language when they call the system to check voice mail. (Additional languages can be purchased separately.)

Subscriber self-enrollment Subscribers personalize their voice mailbox— set up their directory listing, record personal greetings, and set up security codes—through an easy-to-use, touchtone telephone conversation. Because self-enrollment is so simple, subscribers can begin using NEAXMail AD-64 immediately.

- **Directory listing.** Subscribers control whether they want to be listed in the automated attendant directory (see automated attendant features). Subscribers record and spell their name for the directory.
- **Personal greetings.** Subscribers record their individual personal greetings. Callers hear the greeting before they are given an opportunity to leave a message. Subscribers can change the content of their greeting from any touchtone telephone or from their desktop using Mailbox Manager at any time to reflect changes in their schedule.
- **Personal ID.** This number, which is usually based on the subscriber's extension number, identifies subscribers to the voice mail system when they are calling to access their mailbox. The NEAXMail AD-64 allows personal IDs of varying lengths - from one to ten digits.
- **Personal security code.** In addition to a personal ID, each subscriber can also set a security code to safeguard his or her mailbox. The code, which can be any number of digits up to ten, can be changed at any time from any touchtone telephone or by using their Mailbox Manager from their PC.

Subscriber-to-subscriber messaging Subscribers can dial NEAXMail AD-64 directly to leave a message for a fellow subscriber without ringing that extension. In this way, co-workers can exchange routine information and ask questions without disturbing one another. The co-worker can also be dialed directly and, if unavailable, the call is automatically forwarded to the co-worker's voice mail with the caller being identified automatically.

- **Address messages to multiple subscribers.** Subscribers can send a message to many other subscribers at once without creating a group for them.
- **Immediate reply.** NEAXMail AD-64 allows subscribers to reply to a message from another subscriber immediately after hearing it without using touch tones.
- **Message cancellation.** If the recipient has not listened to a message yet, the subscriber sending the message can delete it and record a new message.
- **Multiple message action.** Subscribers can reply to or redirect one message several times. For example, subscribers can listen to part of a message, interrupt it and reply, then start listening again and forward a copy of the message to another subscriber.
- **Message redirection.** NEAXMail AD-64 allows subscribers to forward a copy of a message to another subscriber easily. Subscribers can also record an introduction to the forwarded message.
- **Multiple replies to one message.** Subscribers can respond to old or new messages as many times as necessary.

Text-To-Speech for Exchange®, GroupWise and Lotus Notes (TTS) TTS supports US English, UK English, Australian English, or Dutch however, only one language can be loaded on the system. TTS is supported with all voice board models and has a maximum of 32 TTS sessions.

ViewMail® for Lotus® Notes® (VML) integration is similar to the current ViewMail for Microsoft® Messaging (VMM) application and gives subscribers PC access to their voice messages from their Notes Inbox. 25 Seats of VML are included with any new NEAXMail AD-64 version 2.3 S/W Port License or 2.3 S/W Version Upgrade.

ViewMail® for GroupWise® (VMG) integration is similar to the current ViewMail for Microsoft® Messaging (VMM) application and gives subscribers PC access to their voice messages from their GroupWise Inbox. 25 Seats of VMG are included with any new NEAXMail AD-64 version 2.3 S/W Port License or 2.3 S/W Version Upgrade **Voice sampling rate.** The voice-sampling rate (24Kbps @ 6KHz) provides a high standard of voice messaging.

Voice sampling rate, The voice-sampling rate (24Kbps @ 6KHz) provides a high standard of voice messaging.

Volume control Subscribers can raise or lower the volume of a message by pressing the "5" key during message playback.

Automated Attendant

NEAXMail AD-64 acts as an electronic receptionist, answering and routing incoming calls automatically. Callers hear an opening greeting that gives instructions, information, and options. Using the automated attendant, callers can reach a subscriber directly by dialing the extension number during the opening greeting. The attendant can answer several calls at the same time. It can place callers on hold, inform the caller how many calls are holding ahead of him or her and update this information periodically. It can screen calls, announce the name of the caller, and wait for confirmation from the subscriber before putting the call through. The automated attendant can also be set to answer a specific set of numbers so subscribers and outside callers who know the dedicated number can connect directly to the right person, and the receptionist does not have to waste time transferring calls.

Answer call on first ring NEAXMail AD-64 can be programmed to answer calls on the first ring so incoming calls are handled as quickly as possible.

Caller interviewing NEAXMail AD-64 can interview callers with a series of simple questions (up to 20) through the use of an interview box.

Call screening. NEAXMail AD-64 has the ability to screen calls that are transferred to a subscriber. The subscriber can decide whether or not to take a call on the basis of caller identity. When using this feature, the subscriber hears the caller's name and is asked to press "1" to accept the call or "2" to transfer the call to voice mail. Subscribers can turn call screening on and off by telephone.

Daylight-saving time schedule NEAXMail AD-64 can be programmed to set the clock forward or backward to automatically adjust for daylight-saving time.

Directory listing Callers can enter the first three letters of a subscriber's name (first or last, depending on how the system is configured) and the system routes the call appropriately. Callers are told the extension number for future reference. If the extension is not answered or is busy, NEAXMail AD-64 automatically routes the caller to the subscriber's voice mailbox.

Holiday schedule Holiday dates may be programmed into NEAXMail AD-64 each year so the system automatically runs in night mode on holidays.

Multilingual capability The system can support up to nine different language prompt sets. One system can handle several different languages at once. Language choice can be per port or per user. In addition, callers can choose a language from a language selection box. US English is the default language for all systems, unless otherwise specified.

Available Languages		
American English-US	French Canadian-FC	Mexican Spanish-ES
Argentinean Spanish-AR	German-DE	New Zealand English-NZ
Australian English-AU	Hebrew-HE	Portuguese-PT
Cantonese Chinese-CC	Italian-IT	Swedish-SE
Danish-DK	Latin Am. Spanish-LA	United Kingdom English-UK
Dutch-NL	Madrid Spanish-ED	

Numeric directory assistance NEAXMail AD-64 provides directory assistance for outside callers without letters on their telephone keypad. It can be set up to guide the outside caller through directory assistance using numbers.

One-key dialing this feature allows a caller to press a single touchtone digit instead of a series of touch tones to route their calls. For example, you can route calls quickly to directory assistance, subscriber extensions, transaction boxes, voice-detect boxes, language-select boxes or interview boxes.

Opening greeting NEAXMail AD-64 plays a pre-recorded greeting to callers when it answers a call. The system manager usually records the opening greeting. The greeting typically includes the organization's name; how to reach an extension, operator, and directory; how to send a fax, etc. During the greeting, callers can enter extension numbers, access the directory, or hold for assistance.

Operator assistance Outside callers can press "0" for assistance at any time.

Personal secretary The personal secretary feature enables individual users to have calls transferred to a specified extension when temporarily away from their desk or out of the office. In this way, important calls are handled by a knowledgeable person in the same department rather than by one main operator.

Primary or secondary answering NEAXMail AD-64's automated attendant can be programmed to answer all incoming calls or calls that are not answered by an operator after a pre-set number of rings. Also, the NEAXMail AD-64 can answer all or selected incoming lines.

Public Group (Distribution) Lists. All subscribers on the system can send messages to open groups. A subscriber can be restricted from using open groups. Only subscribers can be members of an open group.

Schedules To accommodate organizations with operating hours that vary from day to day, NEAXMail AD-64 can use up to three different schedules.

Subscriber-controlled call holding. Call holding gives a caller the option to hold for a busy extension to become available. The system can place callers on hold, inform callers of their holding queue position, and update this information periodically. Subscribers can turn call holding on and off by telephone. In addition, certain subscribers can be restricted from being able to turn call holding on and off.

Subscriber-controlled directory listing Individual subscribers can control whether they want to be listed in the directory (by telephone).

Transfer to attendant. Callers with touchtone telephones can press "0" to reach a live operator at any time. Callers using rotary telephones can be transferred to the operator when no touch tones are entered.

Voice detect This feature allows outside callers using rotary telephones to access the voice mail system by speaking, instead of by pressing touch tones. Callers make choices or respond to prompts by saying "yes" or remaining silent for "no," rather than pressing touchtone keys on one-key dialing menus.

Voice detect call holding. Voice detect call holding allows callers without touch tones to hold by saying "yes" instead of pressing "1."

Audiotext

Many offices and organizations want to provide around-the-clock information to clients. NEAXMail AD-64 audiotext features allow an organization to make this information available to callers at all times. The voice messaging system can offer callers menu trees and messages that can vary depending upon the telephone number called or the time of day. Callers use touch tones to select an item or subject that they want to hear. Even multilingual services can be provided. The audiotext recording can be of any length, and there is no limit to the number of audiotext boxes available with the system.

Mailbox Manager

This application brings many voice mail features to your Microsoft Windows-based desktop PC. The graphical user interface (GUI) gives subscribers the ability to set up and control personal mailbox settings in addition to the Telephone User Interface (TUI). With this new tool, subscribers may modify their greetings, security code, notification methods, groups, conversation preferences, and much more.

Record/Edit From a PC or a telephone, subscribers can record their directory name and record and edit their personal greetings, which the NEAXMail AD- 64 plays to incoming callers.

Administer private distribution list (groups) - subscribers oversee their private distribution lists, adding or deleting recipients as necessary.

Change directory listing status - individual subscribers can control whether or not they want to be listed in the directory.

Change telephone password - subscribers can change their telephone password from within the Mailbox Manager.

Change transfer options - subscribers can turn call transfer on or off, and change the extension or telephone number to which incoming calls are transferred. When call transfer is turned off, incoming calls are sent directly to the subscriber's mailbox. This feature is particularly useful for employees who have offices in multiple facilities, frequently work at home, or work as part of a team that requires them to spend a significant amount of time in a colleague's office. By using Mailbox Manager to change call transfer options, callers and subscribers are connected quickly to provide improved customer service and efficient personal administration of call traffic.

Set conversation options - subscribers can specify whether or not they want the default Yes/No or the alternative Menu Mode conversation, whether they want to hear the date and time stamp before or after a message, or if they want the voice mail system to greet them by name when they call in to retrieve their messages.

Set message delivery options - subscribers can establish rules governing message delivery options for the type and urgency of message or frequency of delivery. For example, a subscriber can set the message delivery options so that when he or she receives urgent voice messages on Wednesday, the NEAXMail AD-64 will call the subscriber's home or cellular telephone to deliver them.

Set call screening and holding options - subscribers can set call screening and holding options to handle incoming calls with ease. Two or more of these options may be combined. The following call screening and hold options are available.

- **Announce before connecting caller -** The system plays a beep before transferring the call.
- **Screen Name -** The system asks the caller's name before ringing the extension. The system plays "Call from <caller's name spoken by the caller>" before transferring the call.
- **Ask me if I want to take the call -** The system says "Enter 1 to take the call, or 2 and I'll take a message" and then waits for a response from you before transferring the call.
- **Remember screen name -** The system asks and plays the caller's name as above, and if the caller leaves a message, will append the caller's name to the beginning of the message.
- **Tell me who the call is for** The system says, "Call for <your name>" before transferring the call.

Integration features - NEAXMail AD-64 is capable of delivering the following major features. However, limitations of the NEAX platform, Telco service or network configuration, may make one or more of these features unavailable.

- **Call forward to personal greeting.** This feature makes it easy for callers to leave messages for subscribers who are busy or away from their desks. With this feature, incoming calls routed to an unanswered or busy extension are automatically forwarded directly to a subscriber's voice mailbox where the caller can leave a personal message.
- **Message waiting indication.** NEAXMail AD-64 will light a message waiting lamp, activate a display, or provide a stutter dial tone on the subscriber's extension when new messages have been received.
- **Easy message access.** With this feature, subscribers simply press one button on their telephone set to retrieve new messages from their voice mailboxes without entering a personal ID number. To prevent unauthorized message access, subscribers may also require a security code to be entered before message playback.

- **Live Record.** The feature allows the subscriber to record their telephone conversation and save them as voice messages in their mailbox. Live Record is available on Dterm sets with a properly configured feature button. A programmable “recording beep tone” may be configured where required by law.
- **Constant Message Count.** Allows Dterm display set users to see messaging information on their LCD. The message count is updated to provide old, new and urgent message counts when users are connected to their voice mailbox.
- **Soft Key Mailbox Navigation.** When connected to an IVS2 or IPS system, mailbox subscribers with Dterm Series E and Dterm IP sets have the option of using the four soft key buttons to manage messages and navigate their mailbox menus. Using these visual menu prompts allows the subscriber to quickly select and access various mailbox features.
- **ANI/Caller ID Capture and Display.** When using Enhanced MCI integration Automatic Number Identification (ANI) or Caller ID (CLID) integration displays the telephone number of the outside caller on a desktop PC (using ViewMail, ViewMail for Microsoft Messaging or ViewCall Plus) or the Dterm set’s display during message playback.
- **Immediate disconnect.** NEAXMail AD-64 immediately disconnects when callers hang up, so ports free up immediately.
- **Line resource management controls out-dialing.** This gives priority to incoming call traffic when performing out-dialing operations.

Integration methods – the NEAXMail AD-64 integrates with the NEAX PBX system in one of three ways. The integration method dictates how the NEAX PBX provides call information when forwarding a call to a subscriber’s personal greeting and how it receives message-waiting indication from NEAXMail AD-64. When integrated, both the NEAX PBX and the NEAXMail AD-64 system share information on the origin and destination of calls. It also notifies subscribers when new messages are pending.

- **Direct Digital Integration –** is established using the Intel®/Dialogic® D/42- NE2 voice board(s) installed in the NEAXMail AD-64. Each board emulates up to four Dterm sets and transmits digital switch information between the PBX and the NEAXMail AD-64. The Dialogic board connects to the PBX using one Dterm station port for each NEAXMail AD-64 voice port. The D/42-NE2 board takes a "snapshot" of the feature set display information and identifies the display information for caller and calling party information. This integration also makes the Live Record, Constant Message Count and SoftKey Navigation features possible.
- **Message Center Integration (MCI) -** is a serial data link between the NEAX PBX and NEAXMail AD-64. The voice messaging system and the switch share information about each call by passing data over this serial data link. Message waiting lamp status is sent from the voice messaging system to the telephone system, saving port usage to handle incoming calls. MCI is used in multi-switch networks whenever Common Channel Interoffice Signaling (CCIS) networking is used.
- **Simultaneous MCI & Digital Integration –** is a combination of the previously mentioned integrations and is used whenever multiple NEAX PBXs are networked together via CCIS. With this method the local PBX subscribers may continue to use the Live Record, Constant Message Count, Soft Keys and ANI Capture & Display features. All other integration is performed through the MCI link.
- **Single Point of Entry (SPE)** AIMWorX® Voice Mail SPE is an optional product that lets subscribers manage the most commonly used voice mail functions from their computers. This product provides single-point-of-entry data management between information in the AIMWorX database and the NEAXMail AD-64 voice mail system. You can assign mailboxes, set preferences, and other administrative duties from the Users screen in AIMWorX Manager and then download it to the voice mail system.

Maintenance

The NEAXMail AD-64 is designed to be as easy to maintain as possible. All system hardware requires minimal maintenance. Software maintenance, including backup and restore, is designed to be virtually foolproof.

System Administration Console By using the Windows-based system administration console, NEAXMail AD-64 can be administered from any Windows based PC on your LAN. This new console is intuitively designed and

is accessible via an IP connection, making system administrators more efficient when managing the NEAXMail AD-64 system.

Status Monitor The Status Monitor utility monitors the voice messaging system as it answers and routes calls and it dials phone numbers to deliver messages or turn on message waiting indicators. The Status Monitor utility displays any error messages for ports. The Status Monitor utility can also be used to reset ports if they become unable to answer or place calls.

Disk full warning When the number of minutes of message storage left on the system is equal to or less than a value specified by the system manager, the system asks subscribers to delete unnecessary messages.

On-line reports NEAXMail AD-64 can provide detailed information regarding an organization's incoming telephone calls. The information, generated in reports, can be used to help make decisions about the organization, such as staffing levels, productivity, and telephone equipment needs. Reports can be viewed on the screen or printed on a printer connected to the voice processing system. Reports can also be copied to files and imported into many word processing, database, and spreadsheet programs. There are five types of on-line reports: usage reports, directory reports, busy ports report, call log, and error log.

Usage reports These reports give an indication of how much the voice messaging system is being used over time. These reports can be run for an individual guest, subscriber, extension number, or system ID. Usage of the entire voice messaging system can also be tracked.

- The ***Usage Bar Graph Report*** shows the percentage of each hour that a person or box was using the voice mail system. The bar graph report for the entire system shows the percentage of each hour that the voice mail system's ports were in use. This percentage equals the number of minutes the ports were busy divided by the number of minutes they could have been busy. The number of minutes they could have been busy equals 60 minutes multiplied by the number of ports on the system.
- The ***Usage Table Report*** may be run for an individual guest, subscriber, and extension number ID or system ID and reports the total number of calls by system port and total duration of calls in minutes. A usage table for the entire system shows the total number of calls answered by each system port for each hour of the day and the duration of calls in total number of minutes. The report also includes grand totals for day, night, and entire 24-hour periods.

Directory reports Directory reports show the structure of the system, its subscribers, system IDs and message groups. It is possible to run the report for everyone enrolled in the system, for all extension IDs or for message groups. Directory reports include a subscriber report, extension list and group reports.

Subscriber Reports Subscriber reports list each subscriber and guest enrolled in the system. The report includes each personal ID, the number of new messages waiting, the total number of new and old messages, the date the person last called the voice mail system, the number of days the system saves the subscriber's old and archived messages, and the access code.

Extension List Reports Extension list reports show every subscriber on the system with the corresponding extension number ID and the system IDs for any transaction boxes and interview boxes the subscriber owns. This report does not include guests. For each subscriber, the report lists whether call transfer is currently on or off, the telephone number that calls will be transferred to, the call transfer type, the number of times the extension rings, call transfer options, and whether call holding is on or off.

Group Reports Group reports provide information about message groups in the system. There are three different group reports: List of Message Groups (lists groups), Group Membership (lists members of groups), and Groups Including Person (lists all the message groups of which a subscriber or guest is a member).

Busy port report The busy ports report shows the total number of times all or selected ports were busy. It also shows the average percentage of time the ports were busy during a specific time period. The busy port report can be run for all ports on the system or for a selected group of ports.

Call log The call log provides a record of every call NEAXMail AD-64 answers, dials, or transfers. The call log can be run for the entire system or an individual subscriber.

Error log The error log lists system errors. The report lists each error code, the port on which the error occurred, and the date and time the error occurred. This report can be used to diagnose and solve system problems. The system manager can look up the error code on line and follow the steps to correct the error.

Remote maintenance With NEAXMail AD-64's remote maintenance feature, support and problem solving can be provided without an on-site visit.

Visual Messaging

Visual Messaging is a suite of optional unified messaging and call management modules that can enhance your NEAXMail AD-64 voice messaging system by linking your telephone and your Local Area Network (LAN). Using an intuitive Microsoft Windows interface, the Visual Messaging modules bring calls and messages to your desktop PC and make managing them as easy as clicking a mouse. Instead of listening to calls and messages one-by-one on the telephone, you can visually manage multiple calls or messages on screen and work more efficiently and effectively than ever before. As your organization grows, you can add seats of the different modules. Each module—ViewMail, ViewMail for Microsoft Messaging, ViewMail for Lotus Notes, ViewMail for GroupWise and ViewCall Plus addresses a specific need and can be purchased to customize your NEAXMail AD-64 as your organization's requirements change.

ViewMail

ViewMail organizes messages on screen, making them much easier to manage than on the telephone. ViewMail displays a list of messages that you can quickly scan for important information. You can see who sent the message, the date and time it was sent, the length, and a subject if one was included. When used with Automatic Number Identification (ANI) and Caller ID, or when entered manually through NEAXMail AD-64 call screening feature, the name and number of the person who sent the message is also included. You can sort messages in any order and see details about them at a glance. Intuitive buttons and icons make ViewMail easy to use. You can play and review selected messages, send a message to another subscriber, send a copy of a message to another subscriber, save a message as a WAV file to a desired location and delete messages from your mailbox—all with a click of a mouse. ViewMail can be used on Windows 98, NT 3.51, NT 4.0, 2000, and XP. Here's a quick review of some of these features:

• Archiving messages	• Push button message control
• Call return	• Return receipt
• Copying or pasting messages	• Save as WAV files
• Future message delivery	• Sorting messages
• Message cancel	• VCR-style playback controls
• Message status	• Visual and audio message notification
• Multimedia playback and record	• Volume control
• Password controlled access	

ViewMail for Microsoft Messaging/Lotus Notes/GroupWise

ViewMail for Microsoft Messaging/Lotus Notes/GroupWise gives you the flexibility to send and receive different types of messages interchangeably. With ViewMail for Microsoft Messaging/Lotus Notes/GroupWise, all types of messages—voice, fax*, and e-mail—are collected in your Inbox and made available from a single application ViewMail for Microsoft Messaging/Lotus Notes/GroupWise has all the easy-to-use features of you Inbox, including drop down menus, toolbar buttons, context sensitive help, and tool tips.

With ViewMail for Microsoft Messaging/Lotus Notes/GroupWise, you use intuitive VCR-style buttons for playing messages. You can use the mouse to click on the Rewind, Pause, or Fast Forward buttons to move quickly through messages. A position slider shows where you are in the message, and lets you move

immediately to any part of the message. Using the mouse you can redirect a message to any subscriber, including groups of subscribers. You can also redirect voice over the Internet. ViewMail will save voice messages as WAV files when you attach them to an e-mail message.

ViewMail for Microsoft Messaging/Lotus Notes/GroupWise can be used on Windows 98, NT 4.0, 2000, and XP using Outlook 98, 2000, and 2002. ViewMail for Microsoft Messaging/Lotus Notes/GroupWise gives you some of the same features as the basic ViewMail, including:

• Save as WAV files	• Sorting messages
• Message status	• VCR-style playback controls
• Multimedia playback and record	• Visual and audio message notification
• Password controlled access	• Volume control
• Push-button message control	• Sending messages to the Internet
• Return receipt	• Archiving messages
• Rules-based message handling (with MS Exchange Server)	• Copying and pasting messages to other applications

ViewCall Plus

ViewCall Plus gives you total control over all your incoming, outgoing, external, and internal telephone traffic by managing calls on a PC via a Windows graphical user interface. ViewCall Plus provides live telephone call management on your PC in the same way that unified messaging enables you to manage voice mail and manage data about each caller with the applications—such as Microsoft Office—that you use everyday.

With ViewCall Plus, you simply click a mouse to take calls, send calls to a voice mail box, transfer calls to other extensions, ask callers to hold or hear the caller's name. You can do all this from the desktop PC without picking up the telephone which means you can handle several callers at once without interrupting conversations. Three completely integrated windows let you control every aspect of your telephone traffic. The **Telephone Control** window shows the flow of calls to and from your extension and allows you to interact with callers without interrupting conversations. The **Call Log** window records all call activity. You can sort or print the information in any order, and you can utilize the information to return calls or add contacts to your database by dragging it to those areas with the mouse. The **Contact List** manages data about each caller including multiple telephone numbers and action items and provides you with on-screen information stored in your personal database before connecting with a caller.

For NEAX PBXs enabled with TAPI services (i.e., TAPI Link or PC Telephony Adaptors), ViewCall Plus also offers an **Enhanced mode** that adds powerful new features such as the ability to drag and drop dialing information. To make a call, simply drag the name from the contact list or Call Log to the telephone window and ViewCall Plus will dial the correct number.

ViewCall Plus includes PhoneBASIC® which allows programmers familiar with Microsoft's Visual Basic to customize ViewCall Plus in almost unlimited ways. ViewCall Plus can be configured to access and interact with databases, work with spreadsheets, direct calls according to a set of pre-established rules, handle predictive dialing functions and more. For example, ViewCall Plus can make calls from within an application such as Microsoft Access or Outlook. Here are just a few of the features available with ViewCall Plus:

• Built-in Personal Information Manager (PIM)	• Push-button call management
• Integrates with more than a dozen popular PIM's	• Request hold
• Call log	• Route call
• Call screening	• Sort messages in any order and print
• Call sorting	• TAPI/TSAPI telephone support
• Call status	• Three integrated windows
• Conference calling (Enhanced Mode)	• Visual Basic scripting
• Multiple call handling	
• Drag-and-drop between windows (in Enhanced mode)	• Caller identification (when supported by the telephone system)

NEAXMail IM-16 LX

An enhanced integrated messaging solution; the NEAXMail® IM-16LX is the next generation of internal card-based, voicemail/unified messaging system. It delivers scalable port and message storage capacities with all the features you've come to expect from the most advanced NEC voice messaging products. Designed specifically for the NEAX® 2000 IPS, the internal card set also offers reliability, space and cost-saving advantages.

Up to 490 hours of messages can be stored on the NEAXMail IM-16LX. The integrated messaging solution scales from 4 to 16 ports and expands from 2 to 4 fax ports, making it an ideal messaging choice for the NEAX 2000 business communication platform.

The integrated design of the NEAXMail IM-16LX takes up less space than a standalone messaging solution. It can also save you money. Because the NEAXMail IM-16LX fits inside the NEAX 2000, you don't have to buy line cards normally required for a standalone system. It also uses less energy because it's powered by the NEAX 2000. In the event of a power failure, it will operate off the same battery backup of the NEAX 2000, adding another measure of reliability to your communication system.

Unified Messaging and Call Management

A feature-rich business communications solution, the NEAXMail IM-16LX simplifies message management with convenient access to voice, fax and email messages from any networked PC. Unified Messaging yields fewer misunderstood or lost messages and faster message returns. The result: Employees can prioritize their messages and streamline their workload.

With Voice Messaging, recorded messages can be left and responded to 24x7. Users benefit from many messaging options and the ability to set up their mailboxes to suit individual needs. From a multimedia PC, subscribers can even play and record messages, eliminating the need to pick up a telephone.

Access email messages by phone with the text-to-speech option. Enjoy 24-hour, two-way access to messages without the need for a laptop or modem connection.

Communicating by fax is faster, easier and totally confidential with NEAXMail IM-16LX. Inbound and outbound documents are password protected and can be stored electronically until previewed and printed from any networked desktop PC.

Full Digital Integration

Through its fully customized digital integration, the NEAXMail IM-16LX combines important business communication functions into one comprehensive voice processing system -- a system that can help you increase productivity and improve service dramatically.

Digital integration enables feature options such as Soft key Navigation, Constant Message Count, Live Record, Live Monitor and Mailbox Manager. Soft key Navigation allows you to navigate your Dterm® Series E, I or Dterm IP phone, as well as your NEAXMail IM-16LX voice mail menu options using the LCD screen. Constant Message Count gives you an accurate count of your new or old messages on the LCD screen.

With Live Record, subscribers can record calls at any point during the call and save the recording or forward it to other subscribers. Accurate messages can be passed on to others and phone conversations can be archived for future reference. Setting your mailbox feature options is quick and efficient using the NEAXMail IM-16LX's Web-based Mailbox Manager Application. Subscribers can modify their own mailbox greetings, security codes, notifications, groups, conversation preferences and other settings through the intuitive, graphical user interface accessed with their Microsoft® Internet Explorer browser.

Hospitality Feature Package

NEAXMail IM-16LX offers a version specifically designed for the special needs of the hospitality industry and integrates with over 60 Property Management Systems (PMS).

Personalized guest messaging is available for every room and guests can choose from an extensive list of foreign language prompts. The guest directory allows callers to contact the hotel guests or leave messages without going through an operator. The hospitality package's wake-up feature allows guests to add, delete, change and confirm a wake-up call at any time during their stay. Guest information lines answer routine questions 24 hours a day.

The hospitality package also improves internal communication. The entire hotel staff can be alerted to a message simultaneously. Employees working different schedules can leave messages for each other. Housekeeping, maintenance and room service can be alerted via pager or cell phone while mobile.

Enjoy the Benefits of Integrated Messaging

For simplified message management, take advantage of the NEAXMail IM-16LX's numerous features, convenient access to messages and customizable user-friendly settings. Easy-to-implement, easy-to-maintain and easy-to-install, the NEAXMail IM-16LX is engineered to be a cost-effective and efficient integrated messaging solution for your NEAX 2000.

Basic Features

- Live Record
- Live Monitor
- Soft Keys Functionality
- Web-Based Mailbox Manager
- Browser-based Administration Console
- ANI Capture and Display
- Constant Message Count

- Urgent Message Count
- Date and Time Stamp
- Single Digit Dialing
- Subscriber Self Enrollment
- Special Delivery Options (Future, Urgent, Return Receipt)
- Speed Keys for single button access to functions
- Multiple Personalized Greetings
- Message Editing
- Message Rewind, Pause and Fast Forward
- Message Waiting Lamp Notification
- Out dial Notification to Pager
- Cascade Notification
- Cut-through-paging
- Message Volume Control
- Subscriber Controlled Groups
- Message Cancel and Redirect
- Trunk Mapping for specific trunks to be routed to alternate greetings
- Extension Remapping
- Supervised and Unsupervised Transfer support
- Transfer to Attendant
- User Changeable Voice Prompts
- Reports – Subscriber and System
- Port Independence allows each port to have its own set of parameters to answer calls
- Menu-mode Option
- Alpha and Numeric Directory
- Guest Mailboxes
- Holiday Schedule
- Variable Length Security Codes and Passwords
- Caller Interviewing
- Call Screening
- Call Accept or Reject
- Remote Maintenance
- AIMWorX and MA4000 Integration support
- Fax Detect, Routing and Notification

Additional Features

- Visual Messaging and Call Control Modules: ViewMail, ViewMail for Microsoft Messaging, ViewMail for Lotus Notes, ViewMail for Novell GroupWise, ViewCall Plus, ViewFax.
- Hospitality Feature Package (includes 5 Guest Languages and PMS Integration)
- Active Fax: Fax Mail
- Text-To-Speech Email Reader
- Multiple Languages

Chapter 20 System Documentation

NEC offers a full complement of documents for the *UNIVERGE* NEAX 2000 IPS product line. Technical documentation is available on Compact Disk (CD ROM) or on the WEB through NTAC On-Line (http://www.ilibrary.com/cng/docs.nsf). This section lists all documents included on the Compact Disk (CD ROM).

UNIVERGE NEAX 2000 IPS Documentation List

UNIVERGE **NEAX 2000 IPS Technical Manuals**	
Call Center WorX Business System Manual	Maintenance Manual
Call Center WorX MIS Admin Manual	MATWorX Installation Guide
Call Center WorX MIS Installation Manual	NEAXMail IM 16 LX System Manual
Call Center WorX MIS Reports Manual	OAI System Manual
CCIS System Manual	Office Data Programming
Command Manual	Q-SIG System Manual
Configuration Guide	Q-SIG System Manual (PRT)
Data Interface System Manual	Remote PIM System Manual (Digital Remote)
DM Hardware Installation Guide	Retrofit System Guide
DRS Installation and Configuration Guide	SMDR/MCI/PMS Interface Specification
DTR-1 Dterm Seriesi/ITR-2 DtermIP Install Guide	SNMP Implementation
Feature Programming Manual	System Manual
General Description	Upgrade Guide
INASET Installation Guide	WCS System Manual (PCS)
In-Skin Router Installation Guide	WCS System Manual (PCS) for Series 3300 & up
Installation Procedure Manual	
ISDN System Manual	
ITR-4D-3 DtermIP Installation Guide	

UNIVERGE **NEAX 2000 IPS Features and Specification**	
Business Hotel Features and Specifications	ISDN/Q-SIG Features and Specifications
CCIS Features and Specifications	WCS Features and Specifications
CCWX ACD Features and Specifications	

UNIVERGE **NEAX 2000 IPS User Manuals**	
Dterm Assistant IPS User Guide	MATWorX User Guide
Dterm Series E Agent Console User Guide	MATWorX Studio User Guide
Dterm Series E Supervisor Console User Guide	Power Patch Panel User Guide
Dterm Series E Telephone User Guide	Request for Proposal Guide
Dterm Series I/DtermIP User Guide	SN716 Desk Console User Guide
INASET for IPS User Guide	